# Avaya Agent Deskphone 16CC Administrator Guide

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# Chapter 1: Introduction

## About This Guide

This guide is for personnel who administer Avaya Call Center 5.0 or any component thereof including Avaya Communication Manager, SIP Enablement Services (SES), DHCP, HTTP/HTTPS servers for use by and with Avaya Agent Deskphone 16CC telephones, a Local Area Network (LAN) or a Network Time server.

An Avaya Agent Deskphone 16CC telephone uses Internet Protocol (IP) technology with Ethernet line interfaces and supports the SIP protocol only. Avaya Agent Deskphone 16CC telephones provide support for DHCP, HTTP, and HTTPS over IPv4/UDP, which enhance the administration and servicing of the telephones. These telephones use DHCP to obtain dynamic IP Addresses, and HTTPS or HTTP to download new versions of software or customized settings for the telephones.

**Important:**

This document does not cover call center administration per se, although Chapters 4, 5, and 6 provide some information about what is required. Full documentation for a SIP call center solution is available on the Avaya support Web site, www.avaya.com/support under **Agent Deskphone 16CC**. The document titled *Getting Started with Avaya Call Center 5.0 and Avaya Agent Deskphone 16CC* provides an overview of related tasks and documentation.

Avaya does not provide product support for many of the products mentioned in this document. Take care to ensure that there is adequate technical support available for servers used with any Avaya telephone system. If the servers are not functioning correctly, the telephones might not operate correctly.

# Major Differences Between Avaya Agent Deskphone 16CC and H.323 Telephones in a Call Center

Review this section if your administrative environment includes both SIP and H.323 signaling protocol telephones in your call center. In this section, the phrase "Avaya SIP IP telephone" is used interchangeably with "Avaya Agent Deskphone 16CC."

**General IP Telephony** - Two major protocols handle Voice over IP (VoIP) signaling, Session Initiation Protocol (SIP) and H.323. The two protocols provide connection control and call progress signaling, but in very different ways. These protocols can be used simultaneously over the same network, but in general, no endpoint supports both protocols at the same time. Neither protocol is necessarily superior, but each offers some unique advantages. SIP telephones, for example, do not require centralized call servers, and can route telephone calls when a URL identifies the destination. H.323 telephones leverage the call server’s presence into the potential availability of hundreds of telephone-related features that a standalone SIP telephone cannot provide.

**Signaling** - H.323 call center telephones ship from the factory with H.323 signaling. To use the SIP protocol, applicable H.323 telephones must be appropriately converted and configured. Avaya Agent Deskphone 16CC telephones ship from the factory with SIP signaling, requiring no subsequent conversion.

**Avaya Communication Manager Release** - 16CC telephones are supported only by Communication Manager Release 5.0 and greater. Communication Manager Release 3.0 and up supports 9600 Series IP Telephones (H.323). SIP telephones use Avaya OPS (Outboard Proxy SIP) features on the "trunk" side of Avaya Communication Manager whereas the H.323 (IP) telephones are supported on the "line" side of the Communication Manager.

**Required Servers -** Avaya Agent Deskphone 16CC telephones use two [additional] servers that H.323 telephones do not:

- SIP Proxy server - provided by the SIP Enablement Services (SES) software, and
- Network Time server - which controls time-related parameters.

These servers are not necessarily separate hardware units.

**Features & Functions supported by H.323 9600 Series IP Telephones, Not Supported by SIP:**

This first release of the Avaya Agent Deskphone 16CC does not support the following, non-call center-related features:

- Link Layer Discovery Protocol (LLDP)
- GigE (Gigabit Ethernet)
- Calltype Digit Conversion
- IEEE 802.1X
- Remote Ping & Trace Route
- Web browser
- SBM24 Expansion Modules

**Backup/Restore** - H.323 call center telephones use HTTP to store backup files. Avaya Agent Deskphone 16CC telephones use the Personal Profile Manager (PPM) functionality within SIP Enablement Services (SES) for backup and restore functions.

**Upgrade and Settings Files & System Parameters** - The Avaya Agent Deskphone 16CC has its own unique upgrade script file, **16ccupgrade.txt**, otherwise it uses the same files and parameters as other Avaya SIP/IP telephones. SIP 9600 Series IP Telephones, Avaya Agent Deskphone 16CC telephones, and H.323 9600 Series IP Telephones (and 4600 Series IP Telephones) use the same settings file. Some of the same system parameters are used, however, numerous SIP-specific parameters support SIP operation only.

**Language Support** - This first release of the 16CC telephone supports fewer languages than H.323 telephones. SIP Release 1.0 supports: Canadian French, Parisian French, Latin American Spanish, German, Brazilian Portuguese and English. SIP does not support the Large Text Font for any language. Further, all SIP language files have **.xml** file extensions where H.323 language files have **.txt** file extensions.

**SNMP & MIBs** - Although both 16CC and H.323 telephones support SNMP v2c and have custom Management Information Bases (MIBs), the MIBs are formatted somewhat differently.

**RSVP & VMON** - Avaya SIP IP telephones do not use RSVP (Resource ReSerVation Protocol) or Avaya Voice over IP (VoIP) Monitoring Manager (VMON) software to provide real-time monitoring and historical data of audio quality for VoIP calls. 9600 Series IP Telephones use both RSVP and VMON.

**QoS** - Unlike H.323 telephones, Avaya Agent Deskphone 16CC telephones do not use Avaya Communication Manager to set Quality of Service (QoS). The Avaya SIP IP telephones use the parameters L2QAUD, L2QSIG, DSCPAUD, and DSCPSIG (described in Table 9: Avaya Agent Deskphone 16CC Customizeable System Parameters).

**NAT** - The 16CC does not support Network Address Translation (NAT); 9600 Series IP (H.323) Telephones do support NAT.

# Issue Date

This is the first release of this document, issued for the first time in January, 2008.

# Document Organization

The guide contains the following sections:

| | |
|---|---|
| Chapter 1: Introduction | Provides an overview of this document. |
| Chapter 2: Administration Overview and Requirements | Provides an overview of the administrative process and describes general hardware, software, and operational requirements. |
| Chapter 3: Network Requirements | Describes administrative requirements for your Local Area Network. |
| Chapter 4: Communication Manager Administration | Describes how to administer Avaya Communication Manager to operate with Avaya Agent Deskphone 16CC telephones. |
| Chapter 5: SIP Enablement Services (SES) Administration | Covers SIP Enablement Services (SES) configuration for Avaya Agent Deskphone 16CC telephones. |
| Chapter 6: Server Administration | Describes DHCP and HTTP/HTTPS administration for Avaya Agent Deskphone 16CC telephones. |
| Chapter 7: Telephone Software and Binary Files | Describes telephone software, covers application software downloads, and provides information about the configuration file. |
| Chapter 8: Administering Telephone Options | Describes how to use file parameters and options to administer Avaya Agent Deskphone 16CC telephones. Covers backup and restoration of telephone data. Also describes how to use local procedures to customize a single telephone from the dialpad. |
| Chapter 9: Administering Applications and Options | Describes customizeable application-specific parameters, to provide administrative control of telephone functions and options. |
| Appendix A: Glossary of Terms | Provides a glossary of terms used in this document or which can be applicable to Avaya Agent Deskphone 16CC telephones. |
| Appendix B: Related Documentation | Provides references to Web sites with external documents that relate to telephony in general, and can provide additional information about specific aspects of the telephones. |
| Appendix C: Sample Station Forms and Other Screens | Provides examples of Avaya Communication Manager forms related to system wide and individual telephone administration. |

# Other Documentation

See the Avaya support site at http://www.avaya.com/support for Avaya Agent Deskphone 16CC technical and end user documentation.

See the Avaya support site for documents related to Avaya Call Center 5.0, including:

- *Getting Started with Avaya Call Center 5.0 and Avaya Agent Deskphone 16CC* (Document Number 129214).
- *Avaya Call Center Automated Call Distribution (ACD) Guide* (Document Number 07-600779).
- *Avaya Call Center Release 5.0 Call Vectoring and Expert Agent Selection (EAS) Guide* (Document Number 07-300302).

See Appendix B: Related Documentation for Web sites that list related, non-Avaya documents, such as those published by the Internet Engineering Task Force (IETF) and the International Telecommunication Union (ITU).

# Chapter 2: Administration Overview and Requirements

## Avaya Agent Deskphone 16CC Telephones

The Avaya Agent Deskphone 16CC telephones support the SIP signaling protocol only. Avaya SIP Telephony architecture consists of:

- Avaya Communication Manager (CM), the feature server
- Gateways (also implemented via CM)
- A SES (SIP Enablement Server) which implements the Proxy, Registrar, Event Server, Personal Profile Manager (PPM), System Management Service (SMS), and Location Service components for SIP support

SIP was developed by the IETF. Like H.323, SIP provides for real time audio, video, and data communications transmission over a packet network. SIP uses various messages, or methods, to provide:

- Registration (REGISTER),
- Call signaling (INVITE, BYE)
- Control signaling (SUBSCRIBE, NOTIFY)

The Avaya Agent Deskphone 16CC supports Media Encryption (SRTP) and uses built-in Avaya SIP Certificates for trust management. Trust management involves downloading certificates for additional trusted Certificate Authorities (CA) and the policy management of those CAs. Identity management is handled by Simple Certificate Enrollment Protocol (SCEP) with phone certificates and private keys.

The Avaya Agent Deskphone 16CC telephones are factory-loaded with SIP software, but obtain additional data as part of initial script file administration and initialization during installation. For more information, see Chapter 2 in the *Avaya Agent Deskphone 16CC Installation and Maintenance Guide*. Post-installation, software upgrades automatically download using the proper signaling protocol. For more information, see Chapter 4 in the *Avaya Agent Deskphone 16CC Installation and Maintenance Guide*.

The parameters under which the Avaya Agent Deskphone 16CC needs to operate as a non-call center telephone are summarized as follows:

- Telephone Administration on the Communication Manager (CM) call server, as covered in Chapter 4: Communication Manager Administration.
- Administration on SIP Enablement Services (SES), as covered in Chapter 5: SIP Enablement Services (SES) Administration.
- IP Address management for the telephone, as covered in Chapter 6: Server Administration for dynamic addressing. For static addressing, see the *Avaya Agent Deskphone 16CC Installation and Maintenance Guide*.
- Tagging Control and VLAN administration for the telephone, if appropriate, as covered in Chapter 8: Administering Telephone Options.
- Quality of Service (QoS) administration for the telephone, if appropriate. QoS is covered in QoS on page 27 and QoS on page 36.
- Protocol administration, for example, Simple Network Management Control (SNMP).
- Interface administration for the telephone, as appropriate. Administer the telephone to LAN interface using the PHY1 parameter described in Chapter 3: Network Requirements. Administer the telephone to PC interface using the PHY2 parameter described in “Interface Control” in the *Avaya Agent Deskphone 16CC Installation and Maintenance Guide.*
- Application-specific telephone administration, if appropriate, as described in Chapter 8: Administering Telephone Options.

Table 1 indicates that you can administer system parameters in a variety of ways and use a variety of administrative mechanisms like:

- Maintaining the information on the call server.
- Manually entering the information by means of the telephone dialpad.
- Administering the DHCP server.
- Editing the configuration file on the applicable HTTP or HTTPS file server.
- User modification of certain parameters, when given administrative permission to do so.

**Note:**

Not all parameters can be administered on all administrative mechanisms.

**Table 1: Administration Alternatives and Options for Avaya Agent Deskphone 16CC Telephones**

| Parameter(s) | Administrative Mechanisms | For More Information See: |
|---|---|---|
| Telephone Administration | Avaya Communication Manager and SES | Chapter 4: Communication Manager Administration, Chapter 6: Server Administration, and Appendix B: Related Documentation. |
| IP Addresses | DHCP (strongly recommended) | DHCP and File Servers on page 49, and especially DHCP Server Administration on page 50. |
| | Settings file | Chapter 7: Telephone Software and Binary Files and Chapter 8: Administering Telephone Options. |
| | Manual administration at the telephone | "Static Addressing Installation" in the *Avaya Agent Deskphone 16CC Installation and Maintenance Guide.* |
| Tagging and VLAN | DHCP | DHCP Server Administration on page 50, and Chapter 8: Administering Telephone Options. |
| | Settings file (strongly recommended) | DHCP and File Servers on page 49 and Chapter 8: Administering Telephone Options. |
| | Manual administration at the telephone | "Static Addressing Installation" in the *Avaya Agent Deskphone 16CC Installation and Maintenance Guide*. |
| Network Time Server (NTS) | DHCP Settings file | DHCP Server Administration on page 50 and Network Time Protocol (NTP) Server on page 25. |
| Quality of Service | Settings file | Chapter 8: Administering Telephone Options. |
| Interface | DHCP | DHCP and File Servers on page 49, and Chapter 7: Telephone Software and Binary Files. |
| | Settings file (strongly recommended) | DHCP and File Servers on page 49, and Chapter 7: Telephone Software and Binary Files. |
| | Manual administration at the telephone | "Secondary Ethernet Interface Enable/Disable" in the *Avaya Agent Deskphone 16CC Installation and Maintenance Guide*. |
| Application - specific parameters | DHCP | DHCP and File Servers on page 49, and especially DHCP Server Administration on page 50. Also, Chapter 8: Administering Telephone Options. |
| | Settings file (strongly recommended) | DHCP and File Servers on page 49, and especially HTTP Generic Setup on page 62. Also, Chapter 8: Administering Telephone Options. |

General information about administering DHCP servers is covered in DHCP and File Servers on page 49, and more specifically, DHCP Server Administration on page 50. General information about administering HTTP servers is covered in DHCP and File Servers, and more specifically, HTTP Generic Setup. Once you are familiar with that material, you can administer telephone options as described in Chapter 8: Administering Telephone Options.

# Parameter Data Precedence

If a system parameter is set or changed using multiple administrative mechanisms, the last server/mechanism to provide the parameter has precedence. For example, if the telephone's IP Address is set through DHCP but is changed using the ADDR local dialpad Craft procedure, the IP Address set by the local procedure takes precedence.

The precedence, from lowest to highest, is:

1. DHCP,
2. Settings file,
3. Personal Profile Manager (PPM),

   **Note:**

   Exception: In the case of the parameter SIPDOMAIN, the settings file has a higher precedence than PPM.

4. Manual administration at the telephone, unless the system parameter USE_DHCP is set to 1 (Get IP Address automatically by DHCP), or backup file data obtained through PPM.

# The Administrative Process

The following list depicts administration for a typical Avaya Agent Deskphone 16CC/call center network. Your own configuration might differ depending on the servers and system you have in place.

1. Avaya Communication Manager (5.0 or greater) administered for Avaya Agent Deskphone 16CC telephones.
2. SES (SIP Enablement Services) (5.0 or greater) administered for Avaya Agent Deskphone 16CC telephones.
3. Call Center administration as applicable, for example, Automated Call Distribution (ACD) and Expert Agent Selection (EAS).
4. LAN and applicable servers (file servers, Network Time server) administered to accept the telephones.

5. Telephone software (SIP package) downloaded from the Avaya support site.
6. 46xxsettings file updated with call center-specific, site-specific, and SIP-specific information, as applicable.
7. Avaya Agent Deskphone 16CC telephones installed. For more information, see the *Avaya Agent Deskphone 16CC Installation and Maintenance Guide*.
8. Individual Avaya Agent Deskphone 16CC telephones updated using Craft procedures, as applicable. For more information, see “Local Administrative Procedures” in the *Avaya Agent Deskphone 16CC Installation and Maintenance Guide*.

## Administrative Checklist

Use the following checklist as a guide to system and LAN administrator responsibilities. This high-level list helps ensure that all telephone system prerequisites and requirements are met prior to telephone installation.

**Note:**

One person might function as both the system administrator and the LAN administrator in some environments.

**Table 2: Administrative Checklist**

| Task | Description | For More Information See: |
|---|---|---|
| Network Requirements Assessment | Determine that network hardware is in place and can handle telephone system requirements. | Chapter 3: Network Requirements. |
| Administer Avaya Communication Manager | Verify that the call server is licensed and is administered for Voice over IP (VoIP). | Chapter 4: Communication Manager Administration. |
| | Verify that Avaya Call Center 5.0/CM administration is complete. | TBD (To be determined) |
| | Verify the individual telephones are administered as desired. | Chapter 4: Communication Manager Administration. |
| Administer the Proxy Server | Administer for SIP Enablement Services (SES).<br>Administer for Avaya Call Center 5.0. | *Installing and Administering SIP Enablement Services* (03-600768), available on the Avaya support Web site, http://www.avaya.com/support. |
| DHCP server installation | Install a DHCP application on at least one new or existing PC on the LAN. | Vendor-provided instructions. |

***1 of 2***

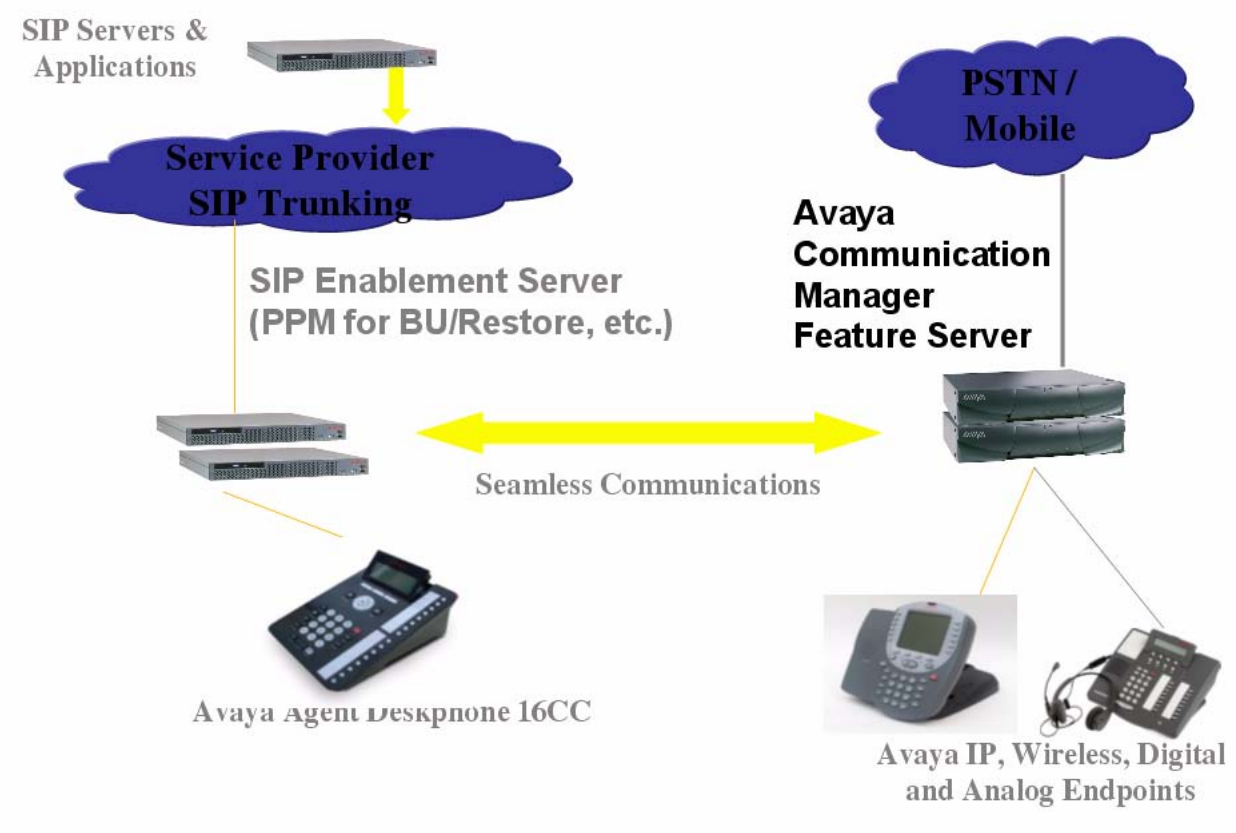
**Table 2: Administrative Checklist (continued)**

| Task | Description | For More Information See: |
|---|---|---|
| Administer DHCP application | Add IP telephone administration to DHCP application. | DHCP Server Administration in Chapter 6: Server Administration. |
| Administer Network Time Server | Set value(s) for Simple Network Time Protocol (SNTP) | Option 42 under DHCP Generic Setup. |
| HTTP/HTTPS server installation | Install an HTTP/HTTPS application on at least one new or existing PC on the LAN. | Vendor-provided instructions. |
| Telephone binary file(s), script file, and settings file installation on HTTP/HTTPS server | Download the Avaya Agent Deskphone 16CC files from the Avaya technical support site. | http://www.avaya.com/support<br><br>Chapter 7: Telephone Software and Binary Files. |
| Modify the settings file | Create/edit the settings file for your specific call center and 16CC telephone system needs using your own tools. | Chapter 7: Telephone Software and Binary Files. |
| Administer telephones locally as applicable | As a Group:<br><br>Individually: | The GROUP System Value on page 68 and the *Avaya Agent Deskphone 16CC Installation and Maintenance Guide.*<br><br>The applicable Craft Local Procedures in the *Avaya Agent Deskphone 16CC Installation and Maintenance Guide.* |
| Installation of telephones in the network | | *Avaya Agent Deskphone 16CC Installation and Maintenance Guide*. |
| Allow user to modify Options, if applicable | Set the following parameters in the settings file:<br>CALL_LOG_ACTIVE<br>ENABLE_CALL_LOG<br>ENABLE_CONTACTS<br>ENABLE_MODIFY_CONTACTS<br>ENABLE_REDIAL<br>ENABLE_REDIAL_LIST<br>PROVIDE_OPTIONS_SCREEN<br>PROVIDE_NETWORKINFO_SCREEN<br>PROVIDE_LOGOUT<br>PROCSTAT | Table 9: Avaya Agent Deskphone 16CC Customizeable System Parameters. |
| Avaya Call Center 5.0 administration | Verify all components are properly administered. | *Avaya Call Center Release 5.0 System Administration Guide.* |

***2 of 2***

# Avaya Agent Deskphone 16CC in the Avaya SIP Call Center Environment

**Figure 1: Avaya Agent Deskphone 16CC in the SIP Call Center Solution**


SIP Servers & Applications
PSTN / Mobile
Service Provider SIP Trunking
Avaya Communication Manager Feature Server
SIP Enablement Server (PPM for BU/Restore, etc.)
Seamless Communications
Avaya Agent Deskphone 16CC
Avaya IP, Wireless, Digital and Analog Endpoints

# Telephone Initialization Process

These steps offer a high-level description of the information exchanged when the telephone initializes and registers. This description assumes that all equipment is properly administered ahead of time. This description can help you understand how the Avaya Agent Deskphone 16CC telephones relate to the routers and servers in your network.

## Step 1: Telephone to Network

The telephone is appropriately installed and powered. After a short initialization process, the telephone identifies the LAN speed and sends a message out into the network, identifying itself and requesting further information. A router on the network receives and relays this message to the appropriate DHCP server.

## Step 2: DHCP Server to Telephone

The DHCP server provides information to the telephone, as described in DHCP and File Servers on page 49. Among other data passed to the telephone is the IP Address of the HTTP or HTTPS server.

## Step 3: Telephone and File Server

The Avaya Agent Deskphone 16CC can download script files, binary files, and settings files from either an HTTP or HTTPS server. The telephone queries the file server, which transmits a script file to the telephone. This script file, at a minimum, tells the telephone which binary file the telephone must use. The binary file is the software that has the telephony functionality.

The telephone uses the script file to determine if it has the proper binary file. If the telephone determines the proper binary file is missing, the telephone requests a binary file download from the file server. The file server then downloads the file and conducts some checks to ensure that the file was downloaded properly. If the telephone determines it already has the proper file, the telephone proceeds as described in the next paragraph without downloading the binary file again.

The telephone checks and loads the binary file, then uses the script file to look for a settings file. The settings file can contain settings you have administered for any or all of the 16CC telephones in your call center/network. For more information about this download process and settings file, see Chapter 7: Telephone Software and Binary Files.

## Step 4: Telephone and the SES Server

In this step, the telephone might prompt the user for an extension and password. The telephone uses that information to exchange a series of messages with SIP Enablement Services (SES), which in turn communicates with Avaya Communication Manager (CM). For a new installation and for full service, the user can enter the telephone extension and the SES password. For a restart of an existing installation, this information is already stored on the telephone, but the user might have to confirm the information. The telephone and SES and SES and CM exchange more messaging. The expected result is that the telephone is appropriately registered and CM call server data such as feature button assignments are downloaded.

For more information about the installation process, see the *Avaya Agent Deskphone 16CC Installation and Maintenance Guide*.

# Error Conditions

Assuming proper administration, most of the problems reported by telephone users are likely to be LAN-based. Quality of Service, server administration, and other issues can impact user perception of telephone performance.

The *Avaya Agent Deskphone 16CC Installation and Maintenance Guide* covers possible operational problems that might be encountered after successful 16CC installation. The User Guide contains guidance for users having problems with specific IP telephone applications.

# Chapter 3: Network Requirements

## Network Assessment

Perform a network assessment to ensure that the network will have the capacity for the expected data and voice traffic, and that it can support for all applications:

- SIP,
- DHCP,
- HTTP/HTTPS, and
- Jitter buffers

Also, QoS support is required to run VoIP on your configuration. For more information, see Appendix B: Related Documentation and the QoS parameters L2QAUD, L2QSIG, DSCPAUD, and DSCPSIG in Table 9: Avaya Agent Deskphone 16CC Customizeable System Parameters.

## Hardware Requirements

To operate properly, you need:

- Category 5e cables designed to the IEEE 802.3af-2003 standard, for LAN powering,
- TN2602 IP Media Processor circuit pack, for Avaya Communication Manager (CM). Sites with a TN2302 IP Media Processor circuit pack are strongly encouraged to install a TN2602 circuit pack.
- TN799C or D Control-LAN (C-LAN) circuit pack.

**Important:**

Avaya IP and SIP telephone firmware requires TN799C V3 or greater C-LAN circuit pack(s). For more information, see the *Communication Manager Software and Firmware Compatibility Matrix* on the Avaya support Web site http://www.avaya.com/support.

To ensure that the appropriate circuit pack(s) are administered on your Communication Manager call server, see Chapter 4: Communication Manager Administration. For more information about hardware requirements in general, see the *Avaya Agent Deskphone 16CC Installation and Maintenance Guide*.

# Server Requirements

Four server types can be configured for Avaya Agent Deskphone 16CC telephones:

- DHCP server
- HTTP or HTTPS server
- SIP Proxy or Registration server
- Network Time Protocol server for SNTP

**Note:**

Avaya Agent Deskphone 16CC telephones and Avaya Call Center 5.0 need SIP Enablement Services (SES) to work properly. The SIP Proxy and Registration servers resides on the SES server. Avaya Communication Manager (CM) is considered a “feature server” behind SES that provides Outboard Proxy SIP (OPS) and advanced SIP telephone features.

While the servers listed provide different functions that relate to the Avaya Agent Deskphone 16CC, they are not necessarily different boxes. For example, DHCP provides network information whereas HTTP provides configuration and binary file management, yet both functions can co-exist on one hardware unit. Any standards-based server is recommended.

For parameters related to Avaya Communication Manager information, see Chapter 4: Communication Manager Administration. For parameters related to DHCP and file servers, see Chapter 6: Server Administration.

**Important:**

The telephones obtain important information from the script files on the server(s) and depend on the binary file for software upgrades. If these servers are unavailable, the telephones will not work and you will need to reset the telephone(s) when the file server is available.

## DHCP Server

Avaya recommends that a DHCP server and application be installed and that static addressing be avoided. Install the DHCP server and application as described in DHCP and File Servers on page 49.

## HTTP/HTTPS Server

Administer the HTTP or HTTPS file server and application as described in HTTP Generic Setup on page 62.

## Network Time Protocol (NTP) Server

SIP telephones require NTP server support to set the time and date, used in system log time stamps and other time/date functions. The NTP server is typically needed by one or more servers within the enterprise. Administration of the NTP server is beyond the scope of this document.

# Required Network Information

Before you administer DHCP and HTTP/HTTPS, as applicable, complete the information in Table 3. If you have more than one router, HTTP/TLS server, and subnetwork mask in your configuration, complete Table 3 for each DHCP server.

The Avaya Agent Deskphone 16CC telephones support specifying a list of IP Addresses for a gateway/router and the HTTP/HTTPS server. Each list can contain up to 255 total ASCII characters, with IP Addresses separated by commas with no intervening spaces. Depending on the specific DHCP application, only 127 characters might be supported.

When specifying IP Addresses for the file server, use either dotted decimal format ("xxx.xxx.xxx.xxx") or DNS names. If you use DNS, the system value DOMAIN is appended to the IP Addresses you specify. If DOMAIN is null, the DNS names must be fully qualified, in accordance with IETF RFCs 1034 and 1035. For more information about DNS, see DHCP Generic Setup on page 52 and DNS Addressing on page 86.

**Table 3: Required Network Information Before Installation - Per DHCP Server**

| | |
|---|---|
| 1. Gateway (router) IP Address(es) | |
| 2. HTTP server IP Address(es) | |
| 3. Subnetwork mask | |
| 4. HTTP server file path (HTTPDIR) | |
| 5. Telephone IP Address range<br>*From*:<br>*To*: | |
| 6. DNS server address(es) | If applicable. |
| 7. HTTPS server address(es) | If applicable. |

The default file server file path is the "root" directory used for all transfers by the server. All files are uploaded to or downloaded from this default directory. In configurations where the upgrade script and binary files are in the default directory, do not use item 4 in Table 3.

As the LAN or System Administrator, you are also responsible for:

- Administering the DHCP server as described in Chapter 6: Server Administration.
- Editing the configuration file on the applicable HTTP or HTTPS file server, as covered in Avaya Agent Deskphone 16CC Telephone Scripts and Binary Files.

# Other Network Considerations

## SNMP

Avaya Agent Deskphone 16CC telephones are fully compatible with SNMPv2c and with Structure of Management Information Version 2 (SMIv2). The telephones respond correctly to queries from entities that comply with earlier versions of SNMP, such as SNMPv1. “Fully compatible” means that the telephones respond to queries directed either at the MIB-II or the read-only Custom MIB.

You can restrict which IP Addresses the telephone accepts SNMP queries from. You can also customize your community string with system values SNMPADD and SNMPSTRING, respectively. For more information, see Chapter 6: Server Administration and Table 9: Avaya Agent Deskphone 16CC Customizeable System Parameters.

**Note:**

SNMP is disabled by default. Administrators must initiate SNMP by setting the SNMPADD and SNMPSTRING system values appropriately.

For more information about SNMP and MIBs in a SIP environment, see *Installing and Administering SIP Enablement Services* (Document Number 03-600768). For general information about SNMP and MIB, see the IETF Web site listed in Appendix B: Related Documentation. The Avaya Custom MIB for the telephones is available for download in *.txt format on the Avaya support Web site at http://www.avaya.com/support.

## Registration and Authentication

An Avaya Agent Deskphone 16CC requires an Outboard Proxy SIP (OPS) extension on Avaya Communication Manager and a login and password on the SES Server to register and authenticate it. Registration is described in the Initialization process, in Step 4: Telephone and the SES Server on page 21. For further information, see *Installing and Administering SIP Enablement Services* (Document Number 03-600768), available on the Avaya support Web site, http://www.avaya.com/support.

## Reliability and Performance

All 16CC telephones respond to a ping or traceroute message sent from Avaya Communication Manager or any other network source. The telephones do not originate a ping or traceroute.

If applicable, the telephones test whether the network Ethernet switch port supports IEEE 802.1D/q tagged frames by ARPing the router with a tagged frame. For more information, see VLAN Considerations on page 82. If your LAN environment includes Virtual LANs (VLANs), your router must respond to ARPs for VLAN tagging to work properly.

## QoS

For more information about the extent to which your network can support any or all of the QoS initiatives, see your LAN equipment documentation. See QoS on page 36 for QoS implications for 16CC telephones.

All telephones provide some detail about network audio quality. For more information see, Network Audio Quality Display on page 27.

## IEEE 802.1D and 802.1Q

For more information about IEEE 802.1D and IEEE 802.1Q and the Avaya Agent Deskphone 16CC telephones, see IEEE 802.1D and 802.1Q on page 36 and VLAN Considerations on page 82. Three bits of the 802.1Q tag are reserved for identifying packet priority to allow any one of eight priorities to be assigned to a specific packet.

- **7:** Network management traffic
- **6:** Voice traffic with less than 10ms latency
- **5:** Voice traffic with less than 100ms latency
- **4:** “Controlled-load” traffic for critical data applications
- **3:** Traffic meriting “extra-effort” by the network for prompt delivery, for example, executive e-mail
- **2:** Reserved for future use
- **0:** The default priority for traffic meriting the “best-effort” for prompt delivery of the network.
- **1:** Background traffic such as bulk data transfers and backups

**Note:**

Priority 0 is a higher priority than Priority 1.

## Network Audio Quality Display

Users can monitor network audio performance while on a call. For more information, see the telephone user guide.

While on a call, the telephones display network audio quality parameters in real-time, as shown in Table 4:

**Table 4: Parameters in Real-Time**

| Parameter | Possible Values |
|---|---|
| Received Audio Coding | ***G.711***, ***G.726A***, or ***G.729***. |
| Packet Loss | No data or a percentage. Late and out-of-sequence packets are counted as lost if they are discarded. Packets are not counted as lost until a subsequent packet is received and the loss confirmed by the RTP sequence number. |
| Packetization Delay | No data or an integer number of milliseconds. The number reflects the amount of delay in received audio packets, and includes any potential delay associated with the codec. |
| One-way Network Delay | No data or an integer number of milliseconds. The number is one-half the value RTCP or SRTCP computes for the round-trip delay. |
| Network Jitter Compensation Delay | No data or an integer number of milliseconds reporting the average delay introduced by the jitter buffer of the telephone. |

The implication for LAN administration depends on the values the user reports and the specific nature of your LAN, like topology, loading, and QoS administration. This information gives the user an idea of how network conditions affect the audio quality of the current call. Avaya assumes you have more detailed tools available for LAN troubleshooting.

## TCP/UDP Port Utilization

The Avaya Agent Deskphone 16CC uses a variety of protocols, particularly TCP, UDP, and TLS to communicate with other equipment in the network. Part of this communication identifies which TCP or UDP or TLS ports each piece of equipment uses to support each protocol and each task within the protocol. For additional TCP/UDP port utilization information as it applies to Avaya Communication Manager, see UDP Port Selection on page 36. For more information on TLS, see Security on page 31.

Depending on your network, you might need to know what ports or ranges are used in the operation of Avaya Agent Deskphone 16CC telephones. Knowing these ports or ranges helps you administer your networking infrastructure.

In Figure 2 and Figure 3:

- The box on the left always represents the Avaya Agent Deskphone 16CC telephone.
- Depending on the diagram, the boxes on the right refer to various pieces of network equipment with which the telephone can communicate.
- Open-headed arrows (for example, ⟶ ) represent the direction(s) of socket initialization.
- Closed-headed arrows (for example, ◀⟶▶ ) represent the direction(s) of data transfer.

- The text the arrows point to identifies the port or ports that the Avaya Agent Deskphone 16CC supports for the specific situation. Brackets identify ranges when more than one port applies. The text indicates any additional qualifications or clarifications. In many cases, the ports used are the ones called for by IETF or other standards bodies.
- Many of the explanations in the diagrams refer to system parameters or options settings. For more information about parameters and settings, see Administering Telephone Options.

**Figure 2: Signaling, Audio and Management Diagram**

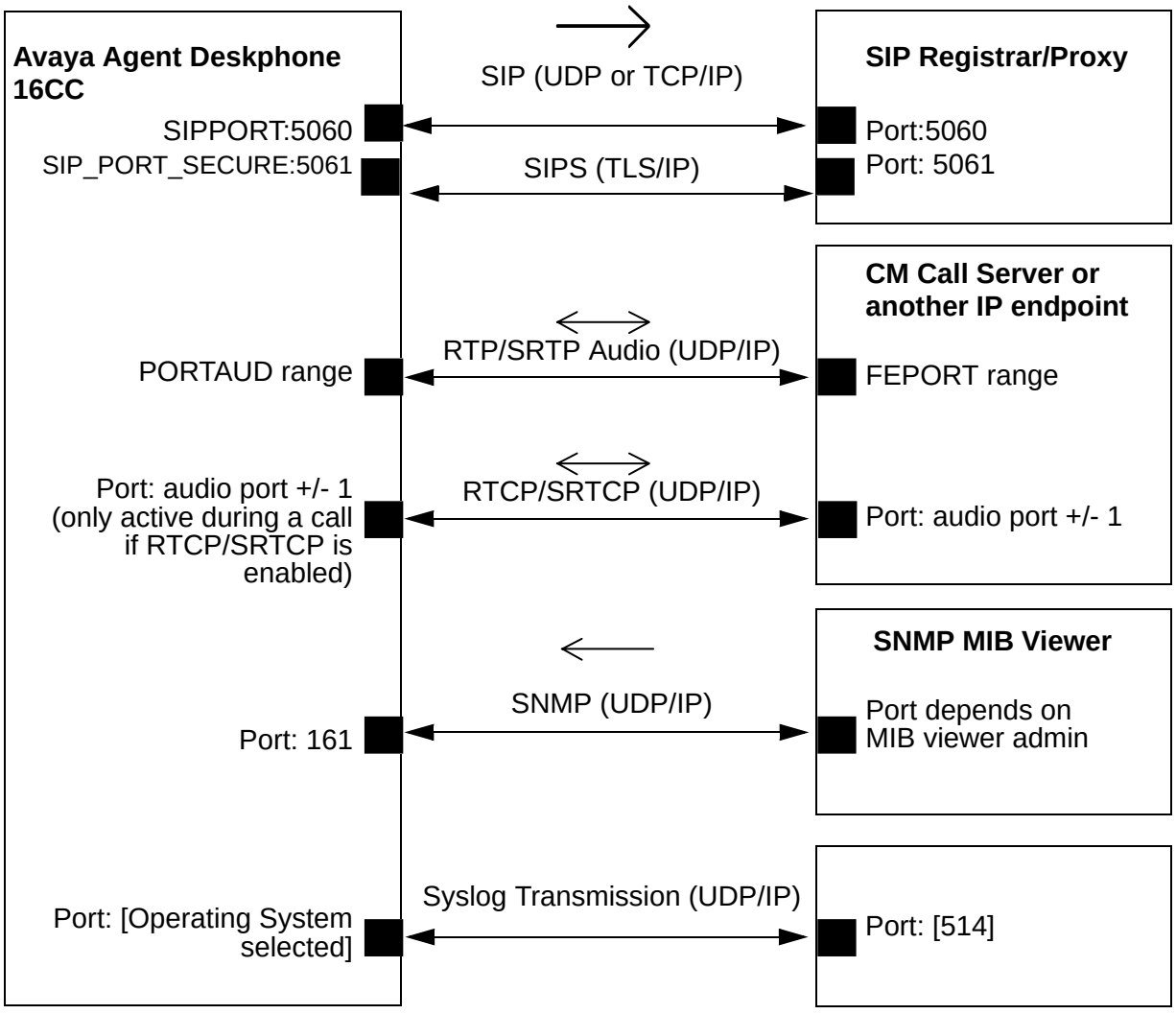
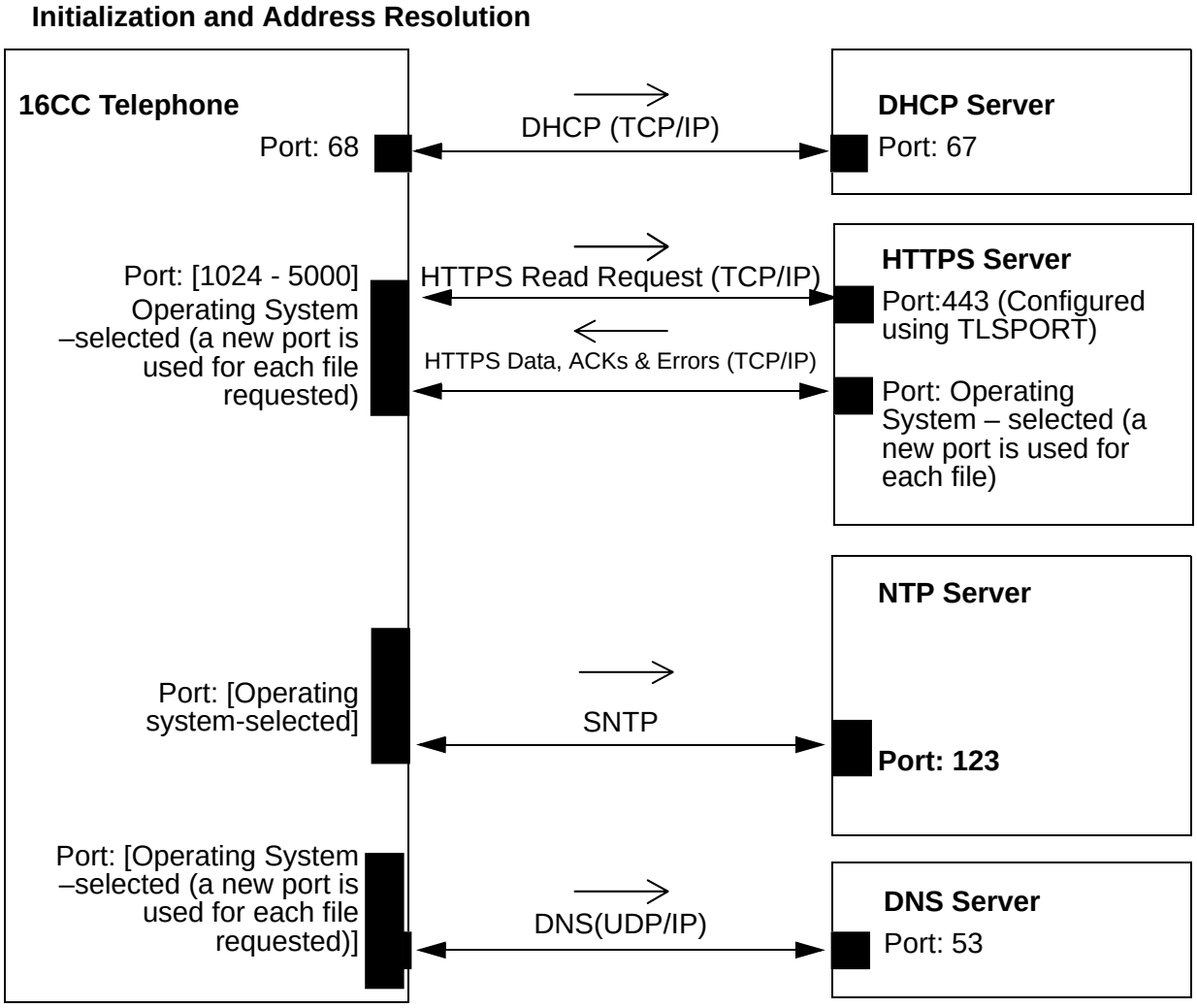


**Figure 3: Initialization and Address Resolution Diagram**



# Security

For information about toll fraud, see the respective call server documents on the Avaya support Web site. The Avaya Agent Deskphone 16CC telephones cannot guarantee resistance to all Denial of Service attacks. However, there are checks and protections to resist such attacks while maintaining appropriate service to legitimate users.

The telephones support Transport Layer Security (TLS) for signaling and for secure communications (SRTP). This mechanism allows the telephone to establish a secure connection to an HTTPS server, in which the upgrade and settings file can reside. This setup adds security over another alternative. The telephone files downloaded during initialization or upgrade are digitally signed, so TLS is not required for this type of file security.

You also have a variety of optional capabilities to restrict or remove how crucial network information is displayed or used. These capabilities are covered in more detail in Chapter 6: Server Administration and include:

- Depending on the SIGSIGNAL parameter, supporting signaling channel encryption while registering, and when registered, with appropriately administered Avaya Communication Manager.
- Restricting the response of the 16CC to SNMP queries to only IP Addresses on a list you specify.
- Specifying an SNMP community string for all SNMP messages the telephone sends.
- Restricting dialpad access to Local Administrative (Craft) Procedures, such as specifying IP Addresses, with a password.
- Restricting dialpad access to Craft Local Procedures to experienced installers and technicians.
- Restricting the end user’s ability to use a telephone Options application to view network data.

## Registration and Authentication

An Avaya Agent Deskphone 16CC requires an Outboard Proxy SIP (OPS) extension on Avaya Communication Manager and a login and password on the SES Server to register and authenticate it. For more information, see the current version of your call server administration manual.

## Subscriptions

Upon successful registration with the Registrar, the 16CC telephone initiates the following subscriptions for the event packages to the server defined in the parameter IPPROXYSRVR_IN_USE:

- Message Waiting. If the MWI event package registration fails, the 16CC still accepts an incoming NOTIFY from the server defined in the parameter SIPPROXYSRVR_IN_USE.
- Avaya-ccs-profile event (if the ENABLE_AVAYA_ENVIRONMENT parameter is 1.)
- Avaya-cm-feature-status event (if the ENABLE_AVAYA_ENVIRONMENT parameter is 1.)
- Dialog event (if the ENABLE_AVAYA_ENVIRONMENT parameter is 1.)

The initial SUBSCRIBE messages are sent with an “expires” value set by the OUTBOUND_SUBSCRIPTION_REQUEST_DURATION parameter. This value might be lowered by the server, depending on the server configuration.

**Note:**

The Avaya Agent Deskphone 16CC does not support presence, and rejects any subscription to the presence event.

After a successful EAS Agent login, the telephone subscribes to the agent status event package. The following event package parameters received in a NOTIFY message are passed to the UI for potential processing:

1. agentEntity
2. nativeEntity
3. currentModeInfo
4. pendingModeInfo
5. errorResponseCode
6. directAgentCallQueued
7. skills
8. messageWaiting
9. logoutReasonInfo
10. auxReasonInfo
11. autoAnswer

Subscriptions are refreshed in accordance with RFC 3265, Section 3.1.4.2.

# Chapter 4: Communication Manager Administration

## Call Server Requirements

Avaya Communication Manager (CM) extends advanced telephony features to SIP telephones via Outboard Proxy SIP (OPS) support. This feature set offers enhanced calling features in advance of SIP protocol definitions and telephone implementations.

Before you perform administration tasks, ensure that the proper hardware is in place, and your call server software is compatible with the 16CC. Avaya recommends:

- the latest CM software,
- the latest SIP telephone firmware.

### Switch Compatibility

The Avaya Agent Deskphone 16CC is supported by Avaya Communication Manager (CM) Release 5.0 and later, as a station type “4620SIPCC.” Administrators will use the 4620SIPCC station screen forms along with the AST OPTIM off-PBX telephone integration screen (“ops” type only) in the same manner as other SIP telephones.

**Note:**

Multiple assignments such as OPS + EC500 is not supported in this Avaya Agent Deskphone 16CC release.

For specific administration instructions about CM/16CC telephones, see Administering Stations on page 43.

## Communication Manager Administrative Requirements

There are several initial CM provisioning tasks that must be performed before administering SIP users. These tasks are described in *SIP Support in Release 4.0 of Avaya Communication Manager Running on the Avaya S8300, S8500, S8500B, S8700, and S8710 Media Server* (Document Number 555-245-206). The tasks administer Communication Manager for SIP Enablement Services (SES) and fall into three categories:

- system-level preparation,
- SIP trunk administration, and
- call routing administration

The sections that follow describe each of these tasks.

## System-Level Preparation Tasks

The system-level preparation tasks include:

- Setting the SIP Trunk capacity on the System Capacity screen.
- Verifying that the IP Trunks field is set to **y** on the System-Parameters Customer-Options screen page 4.
- Verifying that the Maximum Administered SIP Trunks are set correctly on the System Parameters Customer-Options screen page 2.
- Setting the OPS SIP station capacity on the System Parameters Customer Options screen page 1.
- Setting the IP Node name for SES on the IP Node Names screen.
- Entering the IP Address and host name for the administered SES server on the IP Address Mapping screen.
- Setting the Authoritative Domain on the IP Network Region screen.
- Setting the intra- and inter-region IP-IP Direct Audio to yes on the IP Network Region screen.
- Setting the Signaling Group on the Signaling Group screen page 1.
- Setting the License file ISDN/SIP Network Call Redirection field, if applicable.
- Setting the new SIP feature options page "Type of 3PCC Enabled” field appropriately.
- Setting the new license file set customer option “Logged-In SIP EAS Agents” to limit the number of EAS agents who can log in simultaneously using the 16CC. This value limits the number of simultaneous SIP Call Center endpoints for performance reasons.
- Changing system-parameters features to set “Expert Agent Selection (EAS) Enabled?” to “y” (Yes). Define other required options including setting “Minimum Agent-LoginID Password Length:” to other than 0 if password entry is required.
- Adding an agent-loginID xxxxx to assign the agent options and features (including password if needed) as normal.
- Setting up the Avaya Agent Deskphone 16CC station as type = 4620SIPCC and defining the following as appropriate:
  - defining appropriate station options for the 16CC telephone,
  - defining work buttons (Auto-In and/or Manual-In, Aux Work, ACW),
  - defining call appearance/feature buttons,
  - setting auto-answer mode if to be defined by telephone rather than by the Login ID, etc.
- add off-pbx-telephone station-mapping.

## SIP Trunk Administration

SIP trunk administration tasks include:

- Setting the SIP Intercept Treatment and Trunk-to-Trunk Transfer on the System Parameters Features screen page 1.
- Administering Trunk Groups on the Trunk Group screens (pages 1 through 4).
- Assigning public unknown numbering data on the Numbering - Public/Unknown Numbering screen.
- Assigning a SIP phone Set description on Configuration Set screen.
- Changing the SIP trunk group Service Provider Network Call Redirection? field on the Protocol Variations page to "y" (Yes) to allow network call redirection (the default is No) for multi-site call centers.
- Changing the Send Transferring Party Information? field on the Protocol Variations page to "y" (Yes) to make sending transferring party information optional (the default is No).
- Setting UUI (User to User Information) Treatment on the Trunk Group for multi-site call centers if applicable.

## Call Routing Administration

Call routing administration includes:

- Administering Feature Access Codes (FACs) on the Feature Access Code screen.
- Administering the ARS Digit Analysis Table on the ARS Digit Analysis Table screen.
- Administering the Route Pattern on the Route Pattern screen.
- Adding the Route Pattern to the Numbering - Public/Unknown Numbering screen.
- Administering the Proxy Selection Route Pattern on the Locations screen.
- Allowing the system to identify the location of a caller who dials a 911 emergency call from a SIP endpoint on the IP Network Map screen.

The *Administrator Guide for Avaya Communication Manager* (Document Number 03-300509) provides detailed instructions for administering an IP telephone system on Avaya Communication Manager. See Chapter 3 “Managing Telephones,” which describes the process of adding new telephones. Also, you can locate pertinent screen illustrations and field descriptions in Chapter 19 “Screen References” of that guide. You can find this document on the Avaya support Web site.

## IP Interface and Addresses

Follow these general guidelines:

- Define the IP interfaces for each C-LAN and Media processor circuit pack on the switch that uses the IP Interfaces screen. For more information, see *Administration for Network Connectivity for Avaya Communication Manager* (Document 555-233-504).
- On the Customer Options form, verify that the **IP Stations** field is set to "**y**" (Yes). If it is not, contact your Avaya sales representative.

## UDP Port Selection

The Avaya Agent Deskphone 16CC telephones use the next available, even-numbered port, selected from the interval 4000 to 10000. The telephones **cannot** be administered from the Avaya Communication Manager Network Region form to support UDP port selection.

## RSVP and RTCP/SRTCP

Avaya SIP IP Telephones, including the 16CC, support the RTP/SRTP Control Protocol (RTCP/SRTCP). The 16CC telephones do not support RSVP (Resource ReSerVation Protocol).

## QoS

Avaya Agent Deskphone 16CC telephones support both IEEE 802.1D/Q and DiffServ. Other network-based QoS initiatives such as UDP port selection do not require support by the telephones. However, the initiatives contribute to improved QoS for the entire network.

## IEEE 802.1D and 802.1Q

The Avaya Agent Deskphone 16CC can simultaneously support receipt of packets using, or not using, 802.1Q parameters. To support IEEE 802.1D/Q, you can administer the telephones by the value of the following configuration parameters:

- L2Q,
- L2QVLAN,
- L2QAUD, and
- L2QSIG.

## NAT

Avaya Agent Deskphone 16CC telephones do not support Network Address Translation (NAT) interworking.

## DIFFSERV

Type of Service bits 0-5 (also called the Differentiated Services Code Point) are set to the binary equivalent of the decimal number represented by the value of the following configuration parameters:

- DSCPAUD for transmitted audio (RTP, RTCP, SRTP and SRTCP) packets;
- DSCPSIG for transmitted system-specific signaling packets;
- Zero for all other transmitted packets (e.g., DHCP, DNS, HTTP, SNMP, etc.).

Received DSCP information will be ignored.

# Voice Mail Integration

Avaya Agent Deskphone 16CC telephones use the settings file to configure the **Messages** button by setting the system parameter MSGNUM to any dialable string. A MSGNUM example is a standard telephone number the telephone should dial to access your voice mail system, such as AUDIX or Octel.

When the user presses the **Messages** button on the telephone, that number is automatically dialed, giving the user one-touch access to voice mail.

The settings file specifies the telephone number to be dialed automatically when the user presses this button. The command is:

**SET MSGNUM *1234***

where ***1234*** is the Voice Mail extension (CM hunt group or VDN). For more information, see Table 9.

# Call Transfer Considerations

Unlike 9600 H.323 IP Telephones, the 16CC transfer operation is controlled locally by the telephone and is not affected by the settings Abort Transfer?, Transfer Upon Hang-up, and Toggle Swap, on page 7 of the system-parameters features screen.

# Conferencing Call Considerations

Unlike 9600 H.323 IP Telephones, the 16CC conference operation is controlled locally by the phone and is not affected by the settings Abort Conference Upon Hang-up, No Dial Tone Conferencing, Select Line Conferencing, and Toggle Swap, on page 7 of the system-parameters features screen.

# Telephone Administration

Table 5 summarizes the calling features available on Avaya Agent Deskphone 16CC telephones. Some features are supported locally at the telephone, while others are only available with Avaya SIP Enablement Services and Communication Manager with OPS.

The features shown in Table 5 can be invoked at the phone either directly or by selecting a CM-provisioned feature button. Communication Manager automatically handles many other standard calling features via OPS such as call coverage, trunk selection using Automatic Alternate Routing (AAR), or Automatic Route Selection (ARS), Class Of Service/Class Of Restriction (COS/COR), and voice messaging. Details on feature operation and administration can be found in the *Avaya Extension to Cellular and OPS Installation and Administration Guide* (Document Number 210-100-500). The Avaya SIP solution configures all SIP telephones in Communication Manager as OPS.

**Table 5: Avaya Agent Deskphone 16CC Feature Support**

| Feature | Available After Agent Login? | Delivered by Phone? | CM Feature Button/FNU | CM |
|---|---|---|---|---|
| 3-Way Conferencing | | | | X |
| 6-way Conference Bridge | | | | X |
| Automatic Call Back/Cancel | | | FNU | |
| Call Forward All | | | FNU | |
| Call Forward Busy/Don't Answer | | | FNU | |
| Call Forward Deactivation | | | FNU | |
| Call Hold | | X | | |
| Call Management - incoming, outgoing call screening | | | | X |
| Call Park and Retrieve | | | FNU | |
| Call Pick-Up | | | FNU | |
| Calling Number Block/Unblock | | | FNU | |
| Consultation Hold | | X | | |
| Directed Call Pick-Up | | | FNU | |
| Distinctive Alerting | | X | | |
| Find Me | | | | X |
| Group Call Pickup | | | FNU | |
| Last Number Dialed (Redial) | | X | | |
| Message Waiting Indication | | X | | |
| Music on Hold | | | | X |
| Priority Call | | | FNU | |
| Send All Calls/Cancel | | | FNU | |
| Transfer - attended | | X | | |
| Transfer - unattended (one-button transfer) | | | | |
| Transfer to Voice Mail | | | FNU | |
| Whisper Page | No | | FNU | |

# CM/16CC Telephone Configuration Requirements

This section refers to Communication Manager (CM) administration on the Switch Administration Terminal (SAT) or by Avaya Site Administration. The system wide CM form and the particular page that needs to be administered for each feature are provided. These features, which already exist, are not required but are recommended because they optimize the telephone user interface. CM 4.0 or greater is required. For sample Station and other pertinent forms, see Appendix C: Sample Station Forms and Other Screens.

**Table 6: CM/SIP Configuration Requirements**

| Task/Form | Command | Field(s) | Value(s) |
|---|---|---|---|
| IP Network Region | | RTCP Report Period (secs) | SIP telephones have a fixed reporting period. Note that this parameter is only displayed if "Use Default Server Parameters?" is set to "n". |
| IP Network Region | | Authoritative Domain | Make sure that the Authoritative Domain is set to the same value as SIP Domain for Solution. |
| Off-PBX Telephones Station Mapping | change off-pbx-station mapping xxxx | | Bridged call items on this form MUST be “none” or “orig.” |
| Feature - Related System Parameters (page 1) | change system-parameters features | Music/Tone on Hold | This CM setting controls the music on hold capability for all endpoints, including SIP telephones. |
| Feature - Related System Parameters (page 4) | change system-parameters features | Directed Call Pickup | This CM setting controls the availability of directed call pickup. |
| Feature - Related System Parameters (page 4) | change system-parameters features | Extended Group Call Pickup | This CM setting allows a user to answer calls that were directed to another call pickup group. |
| Feature - Related System Parameters (page 17) | change system-parameters features | Whisper Page Tone Given To | This CM setting controls who hears the whisper page tone. |

***1 of 3***

**Table 6: CM/SIP Configuration Requirements (continued)**

| Task/Form | Command | Field(s) | Value(s) |
|---|---|---|---|
| Define the dial plan formats on the Dialplan Analysis Table form | change dialplan analysis | Call Type | Includes all telephone extensions and OPS Feature Name Extensions (FNEs). To define the FNEs for the OPS features listed in Table 5, a FAC must also be specified for the corresponding feature. In a sample configuration, telephone extensions are five digits in length and begin with 3 or 4, FNEs are five digits beginning with 7, and the access codes have various formats as indicated with the Call Type of “fac.” |
| Define the access codes corresponding to the OPS FNEs on the Feature Access Code form | change feature-access-codes | Various fields on pages 1-5 of the form | |
| After defining the FACs, define the FNEs not provisioned by CM feature buttons using the command | change off-pbx-telephone feature-name-extensions | | Used to support both OPS and Extension to Cellular. |
| Set the appropriate service permissions to support OPS features on the Class of Service form | change cos | Varied | y (Yes) or n (No) |

***2 of 3***

**Table 6: CM/SIP Configuration Requirements (continued)**

| Task/Form | Command | Field(s) | Value(s) |
|---|---|---|---|
| Enable applicable calling features on the Class of Restriction form | change cor | Varied | To use the Call Pickup feature, the Can Use Directed Call Pickup and Can Be Picked Up By Call Pickup fields must be set to "y" for the affected stations. Note that Page 3 can be used to implement a form of centralized call screening for groups of stations and trunks |
| Add a station for each SIP phone to be supported using the Station form (page 1) | add station ***xxxxxx*** (where ***xxxxxx*** represents the extension number) | Extension | Assign the same extension as the CM call server extension administered in SIP Enablement Services. See Chapter 5: SIP Enablement Services (SES) Administration and SES documentation for SES configuration information. |
| **Note:** For all other Station Form (screen) page/field information, see Administering Stations on page 43. | | | |
| Stations With Off-PBX Telephone Integration form (page 1) | change off-pbx-telephone station-mapping ***xxxxxx*** where ***xxxxxx*** represents the extension number of the station being configured | Station Extension<br><br>Application<br><br>Dial Prefix<br><br>Phone Number<br><br>Trunk Selection<br><br>Configuration Set | Use to map the Communication Manager extension to the same SIP Enablement Services call server extension. The Application is "OPS." Enter the other appropriate field values, for example, the Trunk Selection value indicates the SIP trunk group. The Configuration Set value can reference a set that has the default settings in Communication Manager. |
| Stations With Off-PBX Telephone Integration form (page 2) | change off-pbx-telephone station-mapping ***xxxxxx*** where ***xxxxxx*** represents the extension number of the station being configured | Call Limit | Change the call limit to match the number of "call-appr" entries in the Add Station form. |

***3 of 3***

# Administering Stations

This section refers to Avaya Communication Manager (CM) administration on the Switch Administration Terminal (SAT) or by Avaya Site Administration. A Station form (pages 1 to 4 and pages 5/6 and up as required) with the Type “4620SIPCC” must be defined for each of the 16CC phones to be used by Call Center agents. Administer the Station form using the guidelines in this section. Sample Station screens are provided in Appendix C: Sample Station Forms and Other Screens.

## Station Screen Field Assignments for the 4620SIPCC (16CC) Station Type

Leave all fields set to their defaults or blank unless they are listed below as assignable.

### Station Screen Page 1 (Access via “add station next”):

| Field: | Assign as: |
|---|---|
| **Extension** | Set this field to the physical number/extension for the phone. |
| **Type** | Set this field to 4620SIPCC, complete the remaining applicable fields on page 1 and submit page 1. Then use "change station xxxx" for remainder of fields and pages, where "xxxx" equals the Extension/physical number of the phone. |
| **Port** | Selected automatically; system-populated. |
| **Name** | Enter the appropriate name for this station, related to where the Call Center station is located. This name should match what is entered for "Name" in the Avaya SES proxy configuration. |
| Fields listed below and fields listed for subsequent Station Screen pages can be set as required. Unless otherwise stated, see the *Administrator Guide for Avaya Communication Manager* (Document 03-300509) for information. | |
| **Lock Messages?** | |
| **Security Code** | |
| **Coverage Path 1** | Set for voice messaging or other hunt group, if available. |
| **Coverage Path 2** | |
| **Hunt-to Station** | |
| **TN** | |
| **COR** | Assign the same value as administered on the Class of Restriction (COR) screen. |
| **COS** | Assign the same value as administered on the Class of Service (COS) screen. |
| **Loss Group** | |
| **Message Lamp Ext** | Enter the extension of the station you want to track with the message waiting lamp. (This value is usually the same value entered as "Extension" on page 1 of the Station form.) |

## Station Screen Page 2:

| Field: | Assign as: |
|---|---|
| **Per Button Ring Control?** | |
| **Bridged Call Alerting?** | Set to "y" if the extension for this SIP telephone will have a "bridged" appearance defined on another non-SIP telephone. Note that no other attributes of the bridged appearance feature apply to SIP telephones (e.g. off-hook indication, bridge-on, etc.). |
| **Active Station Ringing** | Only a single, none, or silent value is supported. |
| **Auto Select Any Idle Appearance?** | Leave this value set to n (no). |
| **Auto Answer** | |
| **Restrict Last Call Appearance?** | By default, the last call appearance is reserved for outgoing calls from a phone. On stations with only three (3) call appearances, set the field to "n" for proper SIP conference and transfer operation. In this mode, all call appearances are available for making or receiving calls. |
| **MWI Served User Type** | Note that MWI (Message Waiting Indicator) applies only to the assigned station extension. |
| **Audix Name** | Enter the name of the voice messaging system administered for the Call Center or the agent's phone, if different. |
| **Per Station CPN - Send Calling Number?** | If CM is configured to always send Caller ID, you can individually block certain stations by setting this field to "n". This field also needs to be set to "n" if you want to use the "Calling Number nblock" FNE. |
| **Coverage After Forwarding?** | This field, with a default of "s" for "system," governs whether an unanswered forwarded call is given CM coverage treatment. |
| **Remote Softphone Emergency Calls** | |
| **Emergency Location Ext** | |
| **Always Use?** | |
| **Direct IP-IP Audio Connections?** | |

## Station Screen Page 3:

| Field: | Assign as: |
|---|---|
| **Conf/Trans on Primary Appearance?** | Avaya recommends that this value be set to "y" for proper transfer or conference operation with bridged appearances. |
| **Bridged Appearance Origination Restricted?** | |

## Station Screen Page 4:

| Field: | Assign as: |
|---|---|
| **Site Data** | Site data fields are display only; they document information related to the station set installation. Note that the Avaya Agent Deskphone supports only a Headset and is not equipped with a speaker. |

# Administering Features

The following buttons can be administered for an Avaya Agent Deskphone 16CC using the Station Screen page 5 (and page 6 if needed):

**Administrable Station Features**

| Feature | Button Name | Administration Notes |
|---|---|---|
| **Call Center Buttons:** | | |
| **Auto-in Mode** | auto-in | Assign if required. If not required or assigned, Manual-in mode button must be assigned. |
| **Manual-in Mode** | man | Assign if required. If not required or assigned, Auto-in mode button must be assigned. |
| **Auxiliary Work** | aux-work | Assign only one Aux-work button. |
| **Auxiliary Work with assigned Reason Code** | aux-work<br>RC: X | Assign one button per unique Reason Code, within the 16 button capacity limits. |
| **After Call Work** | after-call | Optional; assign only one ACW button. |
| **General PBX Buttons (as required):** | | |
| **Audix One-Touch Recording** | audix-rec | |
| **Auto Callback** | auto-cback | |
| **Bridged Call Appearances** | brdg-appr<br>Btn:<br>Extn: | |
| **Call Appearance(s)** | call-appr | Multiple appearances are supported. You must indicate the number of call appearances ("call-appr" buttons) to be supported for this telephone. To support certain transfer and conference scenarios, the minimum number of "call-appr" buttons should be 3. |
| **Call Forwarding (all)** | call-fwd<br>Extn: | Leave the Ext: field blank, as the telephone does not support 3rd party call forwarding. |
| **Call Forwarding (busy/ don't answer)** | cfwd-bsyda | Leave the Ext: field blank, as the telephone does not support 3rd party call forwarding. |
| **Call Park** | call-park | |
| **Call Pickup** | call-pkup | |
| **CPN Block** | cpn-blk | |
| **CPN Unblock** | cpn-unblk | |
| **Directed Call Pickup** | dir-pkup | |
| **Priority Call** | priority | |
| **Send All Calls** | | Leave the Ext: field blank, as the telephone does not support 3rd party send all calls. |
| **Whisper Page** | | This feature cannot be activated when an agent is logged in. |

The following features are administered in an SES configuration file, not through CM:

- Call Unpark
- Extended Call Pickup
- Transfer-to-Voicemail

For additional information about administering Avaya Communication Manager for Avaya Agent Deskphone 16CC telephones, see the following Avaya documents, available on the Avaya Support Web site:

- *Administrator Guide for Avaya Communication Manager* (Document 03-300509).
- *Feature Description and Implementation for Avaya Communication Manager* (Document 555-245-770).
- *Getting Started with Avaya Call Center 5.0 and Avaya Agent Deskphone 16CC* (Document Number 129214).

# Chapter 5: SIP Enablement Services (SES) Administration

## Introduction

SIP Enablement Services (SES) software resides on the SIP Proxy server and provides most of the features and functionality to SIP telephones.This chapter describes using the SES Web browser to configure SES for use with Avaya Agent Deskphone 16CC.

Avaya provides a Web browser to simplify SES administration.

## Using the Web Browser to Configure SES

Sample screens follow this configuration procedure.

1. Set the browser URL to http://IP-address/admin, where **IP-address** is the IP Address of the Avaya SIP Enablement Services Edge or Edge/Home Server.
2. Log in as the administrator “admin” and when prompted, enter the password.

   The main administration screen displays after login.

   **Note:**

   This example administers station 34071 as a SIP endpoint using a 9630 telephone.
3. Click on **Launch Administration Web Interface**.

   The SIP Enablement Services Web interface screen displays.
4. Click **Add** under the **Users** heading on the left side menu.

   The Add User screen displays.
5. . Complete all required fields, indicated by asterisks *.
6. Enter a handle in the **Primary Handle** field. The Primary Handle must be all numeric.
7. Set the **Host** field to the DNS host name of the Avaya SIP Enablement Services Home or Home/Edge server to which the telephone will register.
8. Check the **Add Media Server Extension** checkbox and click **Add**.

   The confirmation screen displays.

9. Click **Continue**.

   The Add Media Server Extension page displays.

10. in the **Extension** field, enter the same extension you entered on page 1 of the Communication Manager Station form. (See for information about Station form entries if necessary).

11. Click **Add**.

    Since the user is being added to Avaya SES Home, the Communication Manager (CM) call server corresponding to the SIP trunk between the Apollo and Home is selected. The confirmation page displays.

12. Click **Continue**.

13. Repeat Steps 4 - 11 for each SIP telephone.

14. When you finish configuring all applicable telephones, click **Update** on the left side menu. This link appears on the current page whenever updates are outstanding, and can be selected at any time to save the administration performed to that point.

# Chapter 6: Server Administration

## Software Checklist

Ensure that you own licenses to use the DHCP, HTTP, and HTTPS server software.

**Note:**

You can install the DHCP and HTTP server software on the same machine.

**Important:**

The firmware in the Avaya Agent Deskphone 16CC reserves IP Addresses of the form **192.168.2.x** for internal communications. The telephone(s) improperly use addresses you specify if they are of that form.

## DHCP and File Servers

Dynamic Host Configuration Protocol (DHCP) minimizes maintenance for the telephone network by removing the need to individually assign and maintain IP Addresses and other parameters for each telephone on the network.

The DHCP server provides the following information to the Avaya Agent Deskphone 16CC:

- IP Address of the telephone(s)
- IP Address of the HTTP or HTTPS server
- IP Address of the NTP (Network Time Protocol) server (using Option 42)
- The subnet mask
- IP Address of the router
- DNS Server IP Address

Administer the LAN so each 16CC telephone can access a DHCP server that contains the IP Addresses and subnet mask.

The telephone cannot function without an IP Address. The failure of a DHCP server at boot time leaves all the affected telephones unusable. A user can manually assign an IP Address to the telephone using the Craft ADDR procedure. When the DHCP server finally returns, the telephone never looks for a DHCP server unless the static IP data is unassigned manually. In addition, manual data entry is an error-prone process.

Avaya recommends that:

- A minimum of two DHCP servers be available for reliability.
- A DHCP server be available when the IP telephone reboots.
- A DHCP server be available at remote sites if WAN failures isolate IP telephones from the central site DHCP server(s).

The file server provides the Avaya Agent Deskphone 16CC with a script file and, if appropriate, new or updated application software. See Step 3: Telephone and File Server on page 20 under Telephone Initialization Process. In addition, you can edit an associated settings file to customize telephone parameters for your specific environment. For more information, see Chapter 8: Administering Telephone Options.

# DHCP Server Administration

This document concentrates on the simplest case of the single LAN segment. Information provided here can be used for more complex LAN configurations.

**Important:**
Before you start, understand your current network configuration. An improper installation will cause network failures or reduce the reliability and performance of your network.

## Configuring DHCP for Avaya Agent Deskphone 16CC

To administer DHCP option 242,if none currently exists, copy Option 176 for your 46xx IP Telephones.You can then either:

- leave any parameters the Avaya Agent Deskphone 16CC does not support for setting via DHCP in option 242 to be ignored, or
- delete unused or unsupported parameters to shorten the DHCP message length.

Only the following parameters can be set in the DHCP site-specific option for 16CC telephones, although most of them can be set in a 46xxsettings.txt file as well.

**Table 7: Parameters Set by DHCP**

| Parameter | Description |
|---|---|
| HTTPDIR | Specifies the path name to prepend to all file names used in HTTP GET operations during startup. (0 to 127 ASCII characters, no spaces.) The command is "**`GET HTTPDIR myhttpdir`**". The path (relative to the root of the HTTP file server) where 96xx telephone files are stored. If an Avaya file server is used to download configuration files over TLS, but a different server is used to download software files via HTTP, set the path of the Avaya server in the DHCP site-specific option, and set HTTPDIR again in the 46xxsettings.txt file with the appropriate path for the second server. HTTPDIR is the path for all HTTP operations. |
| HTTPPORT | Destination port for HTTP requests (0-65535, default is 80). |
| HTTPPROXY | IP or DNS address of the HTTP server for SCEP. 0 to 255 characters in dotted decimal or DNS name format followed by a colon and port number. The colon and port number are optional. If this parameter is not null, this (proxy) transport address is used to set up the HTTP connection as the transport protocol for SCEP. |
| HTTPSRVR | IP Address(es) or DNS name(s) of HTTP file server(s) used to download telephone files. |
| ICMPDU | Controls the extent to which ICMP Destination Unreachable messages are sent in response to messages sent to closed ports so as not to reveal information to potential hackers. The default is 1 (sends Destination Unreachable messages for closed ports used by traceroute). |
| ICMPRED | Controls whether ICMP Redirect messages are processed. The default is 0 (redirect messages are not processed). |
| L2Q | 802.1Q tagging mode. The default is 0 (automatic). |
| L2QVLAN | VLAN ID of the voice VLAN. The default is 0. |
| LOGSRVR | Syslog server IP or DNS address. |
| MTU_SIZE | Maximum transmission unit size. Used to accommodate older Ethernet switches that cannot support the longer maximum frame length of tagged frames (since 802.1Q adds 4 octets to the frame). |
| PHY1STAT | Controls the Ethernet line interface speed. The default is 1 (auto-negotiate). |
| PHY2STAT | Controls the secondary Ethernet interface speed. The default is 1 (auto-negotiate). |
| PROCPSWD | Security string used to access local procedures. The default is 27238. |
| PROCSTAT | Controls whether local procedures are enabled. The default is 0 (enabled). |
| SIPPROXYSRVR | SIP proxy/registrar server IP or DNS address. (0 to 255 characters; zero or one IP Address in dotted decimal or DNS name format, separated by commas without any intervening spaces.) The default is null. |
| SNTPSRVR | List of SNTP server IP or DNS address(es) used to retrieve date and time via SNTP |
| TLSDIR | Used as path name that is prepended to all file names used in HTTPS get operations during initialization (0-127 character string). |
| TLSPORT | Destination TCP port used for requests to https server (0-65535). The default is 443. |
| TLSSRVR | IP Address(es) or DNS name(s) of Avaya file server(s) used to download configuration files.<br>**Note:** Transport Layer Security is used to authenticate the server. |
| VLANTEST | Number of seconds to wait for a DHCPOFFER on a non-zero VLAN. The default is 60 seconds. |

# DHCP Generic Setup

This document is limited to describing a generic administration that works with Avaya Agent Deskphone 16CC telephones. Three DHCP software alternatives are common to Windows operating systems:

- Windows NT® 4.0 DHCP Server
- Windows 2000® DHCP Server
- Windows 2003® DHCP Server

Any other DHCP application might work. It is the responsibility of the customer to install and configure the DHCP server correctly.

DHCP server setup involves:

1. Installing the DHCP server software according to vendor instructions.
2. Configuring the DHCP server with:
   - IP Addresses available for the Avaya Agent Deskphone 16CC telephones.
   - The following DHCP options:
     - **Option 1 - Subnet mask**.
       As described in Table 3, item 3.
     - **Option 3 - Gateway (router) IP Address(es)**.
       As described in Table 3, item 1. If using more than one address, the total list can contain up to 127 total ASCII characters. You must separate IP Addresses with commas with no intervening spaces.
     - **Option 6 - DNS server(s) address list**.
       If using more than one address, the total list can contain up to 255 total ASCII characters. You must separate IP Addresses with commas with no intervening spaces. At least one address in Option 6 must be a valid, non zero, dotted decimal address.
     - **Option 12 - Host Name**.
       Value is **AV*ohhhhhh***, where: *o* is “A” if the OID (first three octets) of the MAC address for the telephone is 00-04-0D. “E” if the OID is 00-09-6E, “L” if the OID is 00-60-1D, and “X” if the OID is anything else and where *hhhhhh* are ASCII characters for the hexadecimal representation of the last three octets of the MAC address for the telephone.
     - **Option 15 - DNS Domain Name**.
       This string contains the domain name to be used when DNS names in system parameters are resolved into IP Addresses. This domain name is appended to the DNS name before the telephone attempts to resolve the DNS address. Option 15 is necessary if you want to use a DNS name for the HTTP server. Otherwise, you can specify a DOMAIN as part of customizing HTTP as indicated in DNS Addressing on page 86.
     - **Option 42 - SNTP Server**.
       This option specifies a list of IP Addresses indicating NTP servers available to the telephone. List servers in the order of preference.The minimum length is 4, and the length must be a multiple of 4.

- **Option 51 - DHCP lease time**.
  If this option is not received, the DHCPOFFER is not be accepted. Avaya recommends a lease time of six weeks or greater. If this option has a value of FFFFFFFF hex, the IP Address lease is assumed to be infinite as per RFC 2131, Section 3.3, so that renewal and rebinding procedures are not necessary even if Options 58 and 59 are received. Expired leases cause the telephones to reboot. Avaya recommends providing enough leases so an IP Address for an IP telephone does not change if it is briefly taken offline.

**Note:**

The DHCP standard states that when a DHCP lease expires, the device should immediately cease using its assigned IP Address. If the network has problems and the only DHCP server is centralized, the server is not accessible to the given telephone. In this case the telephone is not usable until the server can be reached.
Avaya recommends, once assigned an IP Address, the telephone continues using that address after the DHCP lease expires, until a conflict with another device is detected. As Table 9: Avaya Agent Deskphone 16CC Customizeable System Parameters indicates, the system parameter DHCPSTD allows an administrator to specify that the telephone will either:
a). Comply with the DHCP standard by setting DHCPSTD to “1”, or
b). Continue to use its IP Address after the DHCP lease expires by setting DHCPSTD to “0.”
The latter case is the default. If the default is invoked, after the DHCP lease expires the telephone sends an ARP Request for its own IP Address every five seconds.
The request continues either forever, or until the telephone receives an ARP Reply. After receiving an ARP Reply, the telephone displays an error message, sets its IP Address to 0.0.0.0, and attempts to contact the DHCP server again.

- **Option 52 - Overload Option, if desired**.
  If this option is received in a message, the telephone interprets the **sname** and **file** fields in accordance with IETF RFC 2132,
  Section 9.3, listed in Appendix B: Related Documentation.
- **Option 53 - DHCP message type**.
  Value is 1 (DHCPDISCOVER) or 3 (DHCPREQUEST).
- **Option 55 - Parameter Request List**.
  Acceptable values are:
  1 (subnet mask),
  3 (router IP Address[es])
  6 (domain name server IP Address[es])
  7 (log server)
  15 (domain name)
  26 (Interface MTU)
  42 (NTP servers)
  SSON (site-specific option number)
- **Option 57 - Maximum DHCP message size**.

- **Option 58 - DHCP lease renew time**.
  If not received or if this value is greater than that for Option 51, the default value of T1 (renewal timer) is used as per IETF RFC 2131, Section 4.5, listed in Related Documentation.
- **Option 59 - DHCP lease rebind time**.
  If not received or if this value is greater than that for Option 51, the default value of T2 (rebinding timer) is used as per RFC 2131, Section 4.5

The Avaya Agent Deskphone 16CC does not support Regular Expression Matching, and therefore, does not use wildcards. For more information, see Administering Telephone Options on page 69.

In configurations where the upgrade script and binary files are in the default directory on the HTTP server, do not use the HTTPDIR=<path>.

Avaya recommends that you administer DHCP servers to deliver only the options specified in this document. Administering additional, unexpected options might have unexpected results, including causing the IP telephone to ignore the DHCP server.

The SIP Proxy server name and HTTP server name must each be no more than 32 characters in length.

Examples of good DNS administration include:

- Option 6: “***aaa.aaa.aaa.aaa***”
- Option 15: “***dnsexample.yourco.com,zzz.zzz.zzz.zzz***”
- Option 42: "***aaa.aaa.aaa.aaa***"

  Depending on the DHCP application you choose, be aware that the application most likely does not immediately recycle expired DHCP leases. An expired lease might remain reserved for the original client a day or more. For example, Windows NT® DHCP reserves expired leases for about one day. This reservation period protects a lease for a short time. If the client and the DHCP server are in two different time zones, the clocks of the computers are not in sync, or the client is not on the network when the lease expires, there is time to correct the situation.

- The following example shows the implication of having a reservation period: Assume two IP Addresses, therefore two possible DHCP leases. Assume three IP telephones, two of which are using the two available IP Addresses. When the lease for the first two telephones expires, the third telephone cannot get a lease until the reservation period expires. Even if the other two telephones are removed from the network, the third telephone remains without a lease until the reservation period expires.

In Table 8, the Avaya Agent Deskphone 16CC telephone sets the system values to the DHCPACK message field values shown.

**Table 8: DHCPACK Setting of System Values**

| System Value | Set to |
|---|---|
| DHCP lease time | Option #51 (if received). |
| DHCP lease renew time | Option #58 (if received). |
| DHCP lease rebind time | Option #59 (if received). |
| DOMAIN | Option #15 (if received). |
| DNSSRVR | Option #6 (if received, which might be a list of IP Addresses). |
| HTTPSRVR | The **siaddr** field, if that field is non-zero. |
| IPADD | The **yiaddr** field. |
| LOGSRVR | Option #7 (if received). |
| MTU_SIZE | Option #26. |
| NETMASK | Option #1 (if received). |
| ROUTER | Option #3 (if received, which might be a list of IP Addresses). |
| SNTPSRVR | Option #42. |

# Windows NT 4.0 DHCP Server

## Verifying the Installation of the DHCP Server

Use the following procedure to verify whether the DHCP server is installed.

1. Select **Start**-->**Settings**-->**Control Panel**.
2. Double-click the **Network** icon.
3. Verify that **Microsoft DHCP Server** is listed as one of the Network Services on the **Services** tab.
4. If it is listed, continue with the next section. If it is not listed, install the DHCP server.

## Creating a DHCP Scope for the IP Telephones

Use the following procedure to create a DHCP scope for the IP telephones.

1. Select **Start**-->**Programs**-->**Admin Tools**-->**DHCP Manager**.
2. Expand **Local Machine** in the DHCP Servers window by double clicking it until the **+** sign changes to a **-** sign.
3. Select **Scope**-->**Create**.
4. Using information recorded in Table 3: Required Network Information Before Installation - Per DHCP Server:

   Define the **Telephone IP Address Range**.

   Set the **Subnet Mask**.

   To ***exclude*** any IP Addresses you do not want assigned to IP telephones within the **Start** and **End** addresses range:

   a. In the **Exclusion Range Start Address** field, enter the ***first IP Address*** in the range that you want to exclude.

   b. In the **Exclusion Range End Address** field, enter the ***last IP Address*** in the range that you want to exclude.

   c. Click the **Add** button.

   d. Repeat steps a. through c. for each IP Address range to be excluded.

   **Note:**

   Avaya recommends that you provision the Avaya Agent Deskphone 16CC telephones with sequential IP Addresses. Also do not mix Avaya Agent Deskphone 16CC telephones and PCs in the same scope.

5. Under **Lease Duration**, select the **Limited To** option and set the ***lease duration*** to the maximum.
6. Enter a ***sensible name*** for the **Name** field, such as “CM IP Telephones," where CM would represent Avaya Communication Manager.
7. Click **OK**.

   A dialog box prompts you: `Activate the new scope now`?

8. Click **No**.

   **Note:**

   Activate the scope only after setting all options.

## Editing Custom Options

Use the following procedure to edit custom options.

1. Highlight the newly created scope.
2. Select **DHCP Options**-->**Defaults** in the menu.
3. Click the **New** button.
4. In the **Add Option Type** dialog box, enter an appropriate custom option name, for example, “16CCOPTION.”
5. Change the **Data Type Byte** value to **String**.
6. Enter **242** in the **Identifier** field.
7. Click the **OK** button.

   The **DHCP Options** menu displays.
8. Select the **Option Name** for 242 and set the *value string*.
9. Click the **OK** button.
10. For the **Option Name** field, select **003 Router** from the drop-down list.
11. Click **Edit Array**.
12. Enter the ***Gateway IP Address*** recorded in Table 3: Required Network Information Before Installation - Per DHCP Server for the **New IP Address** field.
13. Select **Add** and then **OK**.

## Adding the DHCP Option

Use the following procedure to add the DHCP option.

1. Highlight the scope you just created.
2. Select **Scope** under **DHCP Options**.
3. Select the **242** option that you created from the **Unused Options** list.
4. Click the **Add** button.
5. Select option **003** from the **Unused Options** list.
6. Click the **Add** button.
7. Click the **OK** button.
8. Select the **Global parameter** under **DHCP Options**.
9. Select the **242** option that you created from the **Unused Options** list.
10. Click the **Add** button.
11. Click the **OK** button.

## Activating the Leases

Use the following procedure to activate the leases.

- Click **Activate** under the **Scope** menu.

  The light-bulb icon for the scope lights.

## Verifying Your Configuration

This section describes how to verify that the **96XXOPTIONs** are correctly configured for the Windows NT® 4.0 DHCP server.

**Note:**

Although this configuration represents that for Avaya Agent Deskphone 16CC telephones, the label 96XXOPTIONS or 46XXOPTIONS is suggested. This allows shared use for 4600 Series IP Telephones, 9600 Series IP Telephones (SIP and H.323), and Avaya Agent Deskphone 16CC telephones.

### Verify the Default Option, 242 96XXOPTION

1. Select **Start**-->**Programs**-->**Admin Tools**-->**DHCP Manager**.
2. Expand **Local Machine** in the DHCP servers window by double clicking until the **+** sign changes to a **-** sign.
3. In the DHCP servers frame, click the *scope* for the IP telephone.
4. Select **Defaults** from the **DHCP_Options** menu.
5. In the **Option Name** pull-down list, select **242 96XXOPTION**.
6. Verify that the **Value String** box contains the correct string from DHCP Server Administration.

   If not, update the string and click the **OK** button twice.

### Verify the Scope Option, 242 96XXOPTION

1. Select **Scope** under **DHCP OPTIONS**.
2. In the **Active Options:** scroll list, click **242 96XXOPTION**.
3. Click the **Value** button.
4. Verify that the **Value String** box contains the correct string from DHCP Generic Setup on page 52.

   If not, update the string and click the **OK** button.

### Verify the Global Option, 242 96XXOPTION

1. Select **Global** under **DHCP OPTIONS**.
2. In the **Active Options:** scroll list, click **242 96XXOPTION**.
3. Click the **Value** button.
4. Verify that the **Value String** box contains the correct value from DHCP Generic Setup on page 52. If not, update the string and click the **OK** button.

# Windows 2000 DHCP Server

## Verifying the Installation of the DHCP Server

Use the following procedure to verify whether the DHCP server is installed.

1. Select **Start**-->**Program**-->**Administrative Tools**-->**Computer Management**.
2. Under **Services and Applications** in the Computer Management tree, find **DHCP**.
3. If DHCP is not installed, install the DHCP server. Otherwise, proceed directly to Creating and Configuring a DHCP Scope for instructions on server configuration.

### Creating and Configuring a DHCP Scope

Use the following procedure to create and configure a DHCP scope.

1. Select **Start**-->**Programs**-->**Administrative Tools**-->**DHCP**.
2. In the console tree, click the *DHCP server* to which you want to add the DHCP scope for the IP telephones. This is usually the name of your DHCP server machine.
3. Select **Action**-->**New Scope** from the menu.

   Windows displays the **New Scope Wizard** to guide you through rest of the setup.
4. Click the **Next** button.

   The **Scope Name** dialog box displays.
5. In the **Name** field, enter a name for the scope such as “CM IP Telephones” (where CM would represent Avaya Communication Manager), then enter a brief comment in the **Description** field.
6. When you finish Steps 1 - 5, click the **Next** button.

   The **IP Address Range** dialog box displays.
7. Define the range of IP Addresses used by the IP telephones listed in Table 3: Required Network Information Before Installation - Per DHCP Server. The **Start IP Address** is the first IP Address available to the IP telephones. The **End IP Address** is the last IP Address available to the IP telephones.

   **Note:**

   Avaya recommends not mixing Avaya Agent Deskphone 16CC telephones and PCs in the same scope.

8. Define the **subnet mask** in one of two ways:
   - The number of bits of an IP Address to use for the network/subnet IDs.
   - The subnet mask IP Address.

   Enter only one of these values. When you finish, click the **Next** button.

   The **Add Exclusions** dialog box displays.
9. Exclude any IP Addresses in the range specified in the previous step that you do not want assigned to an IP telephone.
   a. In the **Start Address** field under **Exclusion Range**, enter the ***first IP Address*** in the range you want to exclude.
   b. In the **End Address** field under **Exclusion Range**, enter the ***last IP Address*** in the range you want to exclude.
   c. Click the **Add** button.
   d. Repeat steps a. through c. for each IP Address range that you want to exclude.

   **Note:**

   You can add additional exclusion ranges later by right clicking the **Address Pool** under the newly created scope and selecting the **New Exclusion Range** option.

   Click the **Next** button after you enter all the exclusions.

   The **Lease Duration** dialog box displays.
10. For all telephones that obtain their IP Addresses from the server, enter **30 days** in the **Lease Duration** field. This is the duration after which the IP Address for the device expires and which the device needs to renew.
11. Click the **Next** button.

    The **Configure DHCP Options** dialog box displays.
12. Click the **No, I will activate this scope later** button.

    The **Router** (Default Gateway) dialog box displays.
13. For each router or default gateway, enter the ***IP Address*** and click the **Add** button.

    When you are done, click the **Next** button.

    The **Completing the New Scope Wizard** dialog box displays.
14. Click the **Finish** button.

    The new scope appears under your server in the DHCP tree. The scope is not yet active and does not assign IP Addresses.
15. Highlight the newly created scope and select **Action**-->**Properties** from the menu.
16. Under **Lease duration for DHCP clients**, select **Unlimited** and then click the **OK** button.

    **CAUTION:**

    IP Address leases are kept active for varying periods of time. To avoid having calls terminated suddenly, make the lease duration unlimited.

## Adding DHCP Options

Use the following procedure to add DHCP options to the scope you created in the previous procedure.

1. On the DHCP window, right-click the **Scope Options** folder under the scope you created in the last procedure.

   A drop-down menu displays.

2. In the left pane of the DHCP window, right click the **DHCP Server name**, then click **Set Predefined Options...**.
3. Under **Predefined Options and Values**, click **Add**.
4. In the **Option Type Name** field, enter ***any appropriate name***, for example, “Avaya IP Telephones.”
5. Change the **Data Type** to **String**.
6. In the **Code** field, enter **242**, then click the **OK** button twice.

   The **Predefined Options and Values** dialog box closes, leaving the DHCP dialog box enabled.

7. Expand the newly created scope to reveal its **Scope Options**.
8. Click **Scope Options** and select **Action**-->**Configure Options** from the menu.
9. In the **General** tab page, under the **Available Options**, check the **Option 242** checkbox.
10. In the **Data Entry** box, enter the ***DHCP IP telephone option string*** as described in DHCP Generic Setup on page 52.

    **Note:**

    You can enter the text string directly on the right side of the **Data Entry** box under the ASCII label.

11. From the list in **Available Options**, check option **003 Router**.
12. Enter the ***gateway (router) IP Address*** from the IP Address field of Table 3: Required Network Information Before Installation - Per DHCP Server.
13. Click the **Add** button.
14. Click the **OK** button.

## Activating the New Scope

Use the following procedure to activate the new scope.

1. In the DHCP console tree, click the **IP Telephone Scope** you just created.
2. From the **Action** menu, select **Activate**.

   The small red down arrow over the scope icon disappears, indicating that the scope was activated.

# HTTP Generic Setup

You can store the same application software, script file, and settings file on an HTTP server as you can on a TFTP server. TFTP is not supported for Avaya Agent Deskphone 16CC telephones. With proper administration, the telephone seeks out and uses that material. Some functionality might be lost by a reset if the HTTP server is unavailable. For more information, see DHCP and File Servers on page 49.

**Important:**

The files defined by HTTP server configuration must be accessible from all IP telephones invoking those files. Ensure that the file names match the names in the upgrade script, including case, since UNIX systems are case-sensitive.

**Note:**

Use any HTTP application you want. Commonly used HTTP applications include Apache® and Microsoft® IIS™.

**Important:**

To set up an HTTP server:

- Install the HTTP server application.
- Administer the system parameter HTTPSRVR to the address of the HTTP server. Include this parameter in DHCP Option 242, or the appropriate SSON Option.
- Download the upgrade script file and binary file(s) from the Avaya Web site http://www.avaya.com/support to the HTTP server. For more information, see Contents of the Settings File on page 66.

**Note:**

Many LINUX servers distinguish between upper and lower case names. Ensure that you specify the settings file name accurately, as well as the names and values of the data within the file.

If you choose to enhance the security of your HTTP environment by using Transport Layer Security (TLS), you also need to:

- Install the TLS server application.
- Administer the system parameter TLSSRVR to the address(es) of the Avaya HTTP server.

# Chapter 7: Telephone Software and Binary Files

## General Download Process

The Avaya Agent Deskphone 16CC downloads script files, binary files, and settings files from either an HTTP or HTTPS server. The HTTPS server applies only if the server supports Transport Layer Security (TLS) encryption.

**Note:**

The script files, binary files, and settings files discussed in this chapter are identical for HTTP and HTTPS servers. The generic term “file server” refers to both “HTTP server” and “HTTPS server.”

The file downloading process is the same for both servers, except that when you use an HTTPS server, a TLS server is contacted first. The telephone queries the file server, which transmits a script file to the telephone. The script file tells the telephone which binary file the telephone must use. The binary file is the software that has the telephony functionality, and is easily updated for future enhancements. In a newly installed telephone, the binary file might be missing. In a previously installed telephone, the binary file might not be the proper one. In both cases, the telephone requests a download of the proper binary file from the file server. The file server downloads the file and conducts some checks to ensure that the file was downloaded properly. If the telephone determines it already has the proper file, the telephone proceeds to the next step without downloading the binary file again.

After checking and loading the binary file, the Avaya Agent Deskphone 16CC telephone, if appropriate, uses the script file to look for a settings file. The settings file contains options you have administered for any or all of the IP Telephones in your network. For more information about the settings file, see Contents of the Settings File on page 66.

## Software

When shipped from the factory, an Avaya Agent Deskphone 16CC does not contain sufficient software for registration and operation. When the telephone is first plugged in, a software download from an HTTP or HTTPS server starts to give the phone its proper functionality. For software upgrades, SIP Enablement Services (SES) provides the capability for a remote reboot of the 16CC. As a result, the telephone automatically starts reboot procedures. If new software is available on the server, the telephone downloads it as part of the reboot process. The *Avaya Agent Deskphone 16CC Installation and Maintenance Guide* covers upgrades to a previously installed telephone and related information.

# Avaya Agent Deskphone 16CC Telephone Scripts and Binary Files

## Choosing the Right Binary File and Upgrade Script File

The software releases containing the files needed to operate Avaya Agent Deskphone 16CC telephones are bundled together. You download this self-extracting executable file to your file server from the Avaya support Web site. To get the correct files:

- Go to http://www.avaya.com/support.
- Search for **Agent Deskphone 16CC**
- Select **Downloads**

The 16CC bundle contains:

- An upgrade script file, **16ccupgrade.txt**, which allows you to upgrade to new software releases and new functionality without having to replace 16CC telephones. The upgrade script tells the telephone whether a software upgrade is needed. All Avaya Agent Deskphone 16CC telephones attempt to read this file whenever they reset. The upgrade script file is also used to point to the settings file.
- parameter settings and values that customize the telephones for your enterprise. One settings file is used for all your Avaya IP Telephones.
- Binary files for all current Avaya Agent Deskphone 16CC telephones.
- Other useful information such as a ReadMe file.

In addition to the upgrade script, application files, and Read Me file you need the latest binary code the Avaya Agent Deskphone 16CC telephones use. All of these required files are in the self-extracting executable file that comes in both zipped and unzipped format.

## Upgrade Script File

An upgrade script file named **16ccupgrade.txt**, tells the telephone whether the telephone needs to upgrade software. The Avaya Agent Deskphone 16CC telephones attempt to read this file whenever they reset. The upgrade script file also points to the settings file.

You download the upgrade script file, sometimes called the “script file,” from http://www.avaya.com/support. This file allows the telephone to use default settings for customer-definable options. All files must reside in the same directory.

## Settings File

The settings file contains the option settings you need to customize the Avaya Agent Deskphone 16CC telephones for your enterprise.

**Note:**

Use one settings file, **46xxsettings.txt**, for all your Avaya IP Telephones. The settings file includes the Avaya Agent Deskphone 16CC telephones covered in this document. The settings file also includes 9600 Series (H.323) IP Telephones, 4600 Series IP Telephones, and 1600 Series IP Telephones as covered in their respective administrator guides.

Avaya recommends that the settings file have the extension ***.txt**. You can change some settings with DHCP or, in some cases, from the telephone dialpad using local administrative (Craft) procedures.

The settings file can include any of five types of statements, one per line:

- Comments, which are statements with a “#” character in the first column.
- Tags, which are comments that have exactly one space character after the initial #, followed by a text string with no spaces.
- **`Goto`** commands, of the form **GOTO *tag.*** **`Goto`** commands cause the telephone to continue interpreting the configuration file at the next line after a **# *tag*** statement. If no such statement exists, the rest of the configuration file is ignored.
- Conditionals, of the form **IF *$name* SEQ *string* GOTO *tag***. Conditionals cause the **`Goto`** command to be processed if the value of ***name*** is a case-insensitive equivalent to ***string***. If no such ***name*** exists, the entire conditional is ignored. The only system values that can be used in a conditional statement are: BOOTNAME, GROUP, and SIG.

- **SET** commands, of the form **SET *parameter_name value***. Invalid values cause the specified value to be ignored for the associated ***parameter_name*** so the default or previously administered value is retained. All values must be text strings, even if the value itself is numeric, a dotted decimal IP Address, and so on.

  **Note:**

  Enclose all data in quotation marks for proper interpretation.

The upgrade script file Avaya provides includes a line that tell the telephone to **GET *46xxsettings.txt***. This lines causes the telephone to use HTTP to attempt to download the file specified in the **GET** command. If the file is obtained, its contents are interpreted as an additional script file. That is how your settings are changed from the default settings. If the file cannot be obtained, the telephone continues processing the upgrade script file.

If the configuration file is successfully obtained but does not include any setting changes the telephone stops using HTTP. This happens when you initially download the script file template from the Avaya support Web site, before you make any changes. When the configuration file contains no setting changes, the telephone does not go back to the upgrade script file.

**Note:**

Language files are downloaded using HTTP. For more information, see Language Selection on page 87.

Avaya recommends that you do **not** alter the upgrade script file. If Avaya changes the upgrade script file in the future, any changes you have made will be lost. Avaya recommends that you use the ***46xxsettings*** file to customize your settings instead. However, you can change the settings file name, if desired, as long as you also edit the corresponding **GET** command in the upgrade script file.

For more information on customizing your settings file, see Contents of the Settings File.

## Contents of the Settings File

After checking the application software, the telephone looks for a 46xxsettings file. This file is where you identify non-default option settings, application-specific parameters, and so on. You can download a template for this file from the Avaya support Web site. An example of what the file might look like follows.

**Note:**

The following is intended only as a simple example. Your settings will vary from the settings shown. This sample assumes specification of a DNS Server, identifying SIP-specific settings, and setting the time/date.

SET DNSSRVR=”dnsexample.yourco.com”

SET SIPPROXYSRVR=192.168.1.110

SET SIPSIGNAL=”1” (TCP)

SET ENABLE_PRESENCE=”1” (show presence icons)

SET SIPDOMAIN=”domain name”

SET SNTPSRVR=192.168.1.111

GMTOFFSET “-5:00”

DSTOFFSET “1”

DSTSTART “2SunMar2L” (second Sunday in March at 2 am Local time)

DSTSTOP “1SunNov2L” (first Sunday in November at 2 am Local time)

Note that the DSTSTART and DSTSTOP parameters reflect the new 2007 Daylight Savings Time values for North America

See Chapter 8: Administering Telephone Options for details about specific values. You need only specify settings that vary from defaults, although specifying defaults is harmless.

VLAN separation controls whether or not traffic received on the secondary Ethernet interface is forwarded on the voice VLAN and whether network traffic received on the data VLAN is forwarded to the telephone. Add commands to the 46xxsettings.txt file to enable VLAN separation. The following example assumes the data VLAN ID is “yyy” and the data traffic priority is “z”:

```
SET VLANSEP 1
SET PHY2VLAN yyy
SET PHY2PRIO z
```

**Note:**

Also configure the network switch so that 802.1Q tags are not removed from frames forwarded to the telephone.

# The GROUP System Value

You might have different communities of users, all of which have the same telephone model, but which require different administered settings. For example, you might want to restrict Call Center agents from being able to Logoff, which might be an essential capability for “hot-desking” associates. We provide examples of the group settings for each of these situations later in this section.

Use the GROUP system value for this purpose:

1. identify which telephones are associated with which group, and designate a number for each group. The number can be any integer from 0 to 999, with 0 as the default, meaning your largest group is assigned as Group 0.
2. At each non-default telephone, instruct the installer or user to invoke the GROUP Craft Local procedure as specified in the *Avaya Agent Deskphone 16CC Installation and Maintenance Guide* and specify which GROUP number to use. The GROUP System value can only be set on a phone-by-phone basis.
3. Once the GROUP assignments are in place, edit the configuration file to allow each telephone of the appropriate group to download its proper settings.

Here is an example of the configuration file that accounts for two group settings:

```
IF $GROUP SEQ 1 goto GROUP1
IF $GROUP SEQ 2 goto GROUP2
{specify settings unique to Group 0}
goto END

# GROUP1
{specify settings unique to Group 1}
goto END

# GROUP2
{specify settings unique to Group 2}

# END
{specify settings common to all Groups}
```

# Chapter 8: Administering Telephone Options

## Administering Options for Avaya Agent Deskphone 16CC Telephones

This chapter explains how to change parameters by means of the DHCP or HTTP servers. In all cases, you are setting a system parameter in the telephone to a desired value. Table 9 lists:

- the parameter names,
- their default values,
- the valid ranges for those values, and
- a description of each one.

For DHCP, the DHCP Option sets these parameters to the desired values as discussed in DHCP and File Servers on page 49. For HTTP, the parameters in Table 9 are set to desired values in the script file. For more information, see Contents of the Settings File on page 66.

Avaya recommends that you administer options using script files. Some DHCP applications have limits on the amount of user-specified information. The administration required can exceed those limits for the more full-featured telephone models.

You might choose to completely disable the capability to enter or change option settings from the dialpad. You can set the system value, PROCPSWD, as part of standard DHCP/HTTP administration. If PROCPSWD is non-null and consists of 1 to 7 digits, a user cannot invoke any local options without first entering the PROCPSWD value on the Craft Access Code Entry screen. For more information on craft options, see the *Avaya Agent Deskphone 16CC Installation and Maintenance Guide*.

**Important:**

PROCPSWD is likely stored on the server “in the clear” and is sent to the telephone in the clear. Therefore, do not consider PROCPSWD as a high-security technique to inhibit a sophisticated user from obtaining access to local procedures.

Administering PROCPSWD limits access to all local procedures, including VIEW. VIEW is a read-only Craft option that allows review of the current telephone settings.

**Table 9: Avaya Agent Deskphone 16CC Customizeable System Parameters**

| Parameter Name | Default Value | Description and Value Range |
|---|---|---|
| AGCHEAD | 1 | Automatic Gain Control status for headset. Values are 0=disabled, 1=enabled. |
| AGENTTONE | 1 | Call center agent confirmation tone. ID of tone played when a call center agent successfully logs in, as described in Agent Confirmation Tones on page 90. |
| AUDASYS | 3 | Globally controls audible alerting. Values range from 0 through 3. Value 0 or 2=audible alerting off. Value 1 or 3=audible alerting on. |
| AUDIOENV | 0 | Audio environment selection index. Values range from 0 through 191. |
| AUDIOSTHD | 0 | Headset sidetone setting. Values are:<br>0 = Nominal<br>1 = -3dB below nominal<br>2 = -9dB below nominal<br>3 = -15dB below nominal<br>4 = -30dB below nominal (essentially no sidetone)<br>5 = 10dB above nominal |
| AUTH | 0 | Authentication flag for settings file download. Values are:<br>0=secure setting file download is not required<br>1=secure setting file download is required. |
| BAKLIGHTOFF | 120 | Number of minutes without display activity to wait before turning off the backlight. Values range from zero (never turn off) through 999 minutes (16.65 hours). |
| CONFIG_ SERVER | " "(Null) | Address of Avaya configuration server. When used, this parameter is set to the PPM server address. Format is dotted decimal or DNS format, separated by commas, with no spaces (0-255 ASCII characters, including commas, optionally followed by a colon and port number). |
| COUNTRY | USA | Country of operation for specific dial tone generation. This parameter also controls network progress tones. |
| DATEFORMAT | %m/%d/%y | Formatting string defining how to display the date in the top line and the call log. |
| DAYLIGHT_SAVING_ SETTING_MODE | 2 | Controls daylight saving setting. Values are:<br>0=daylight saving time is deactivated (no offset to local time).<br>1=daylight saving time is activated (offset to local time as configured in "DSTOFFSET") .<br>2=the device switches automatically to daylight saving time and back according to the contents of "DSTSTART" and "DSTSTOP." |

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**Table 9: Avaya Agent Deskphone 16CC Customizeable System Parameters (continued)**

| Parameter Name | Default Value | Description and Value Range |
|---|---|---|
| DHCPSTD | 0 | DHCP Standard lease violation flag. Indicates whether to keep the IP Address if there is no response to lease renewal. If set to “1” (No) the telephone strictly follows the DHCP standard with respect to giving up IP Addresses when the DHCP lease expires. If set to “0” (Yes) the telephone continues using the IP Address until it detects reset or a conflict (see DHCP Generic Setup). |
| DNSSRVR | 0.0.0.0 | Text string containing the IP Address of zero or more DNS servers, in dotted-decimal format, separated by commas with no intervening spaces (0-255 ASCII characters, including commas). |
| DOMAIN | " " (Null) | Text string containing the domain name to be used when DNS names in system values are resolved into IP Addresses. Valid values are 0-255 ASCII characters. |
| DSCPAUD | 46 | Differentiated Services Code Point for audio. Values range from 0 to 63. |
| DSCPSIG | 34 | Differentiated Services Code Point for signaling. Values range from 0 to 63. |
| DSTOFFSET | 1 | Used for daylight saving time calculation in hours. Values range from 0 to 2. |
| DSTSTART | 2Sun Mar2L | Used to identify start date for automatic change to Daylight Saving Time. Default string length with a format of either *odddmmmht* or *Dmmmht*, where:<br>*o* = one character representing an ordinal adjective of "1" (first), "2" (second), "3" (third), "4" (fourth) or "L" (last).<br>*ddd* = 3 characters containing the English abbreviation for the day of the week.<br>*mmm* = 3 characters containing the English abbreviation for the month.<br>*h* = one numeric digit representing the time to make the adjustment, exactly on the hour at hAM (0h00 in military format), where valid values of h are "0" through "9"<br>*t* = one character representing the time zone relative to the adjustment where "L" is local time and U is universal time.<br>*D* = one or two ASCII digits representing the date of the month from "1" or "01" to "31", or the character "L", which means the last day of the month). |

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**Table 9: Avaya Agent Deskphone 16CC Customizeable System Parameters (continued)**

| Parameter Name | Default Value | Description and Value Range |
|---|---|---|
| DSTSTOP | 1SunNov2L | Used to identify stop date for automatic change to Daylight Saving Time. Default string length with a format of either *odddmmmht* or *Dmmmht*, where:<br>*o* = one character representing an ordinal adjective of "1" (first), "2" (second), "3" (third), "4" (fourth) or "L" (last).<br>*ddd* = 3 characters containing the English abbreviation for the day of the week .<br>*mmm* = 3 characters containing the English abbreviation for the month .<br>*h* = one numeric digit representing the time to make the adjustment, exactly on the hour at hAM (0h00 in military format), where valid values of h are "0" through "9".<br>*t* = one character representing the time zone relative to the adjustment where "L" is local time and U is universal time.<br>*D* = one or two ASCII digits representing the date of the month from "1" or "01" to "31", or the character "L", which means the last day of the month). |
| DTMF_PAYLOAD_TYPE | 120 | RTP dynamic payload used for RFC 2833 signaling. Range is 96 to 127. |
| ENABLE_CALL_LOG | 1 | Enable or disable complete Call Log application. If disabled no calls are logged, screens related to Call Log are not displayed to user, and menu items of User Interface to set Call Log options are not displayed. Values are 0=disabled; 1=enabled. |
| ENABLE_CONTACTS | 1 | Enable or disable complete Contact application. If disabled no contacts are downloaded during initialization from PPM, screens related to Contacts application are not displayed to user, and menu items of the User Interface to set Contacts options are hidden. Values are 0=disabled; 1=enabled. |
| ENABLE_EARLY_MEDIA | 1 | Flag that indicates if SIP early is enabled. If enabled and 18x progress message includes early SDP, that information is used to open a VoIP channel to the far-end before the call is answered. Values are 0=disabled; 1=enabled. |
| ENABLE_G711A | 1 | Enable or disable G711A codec capability of the phone. If the parameter is set to 1, the phone includes G711A capability in an outbound INVITE request, and accepts G711A when received in an incoming INVITE request. Values are 0=disabled; 1=enabled. |
| ENABLE_G711U | 1 | Enable or disable G711U codec capability of the phone. If the parameter is set to 1, the phone includes G711U capability in an outbound INVITE request, and accepts G711U when received in an incoming INVITE request. Values are 0=disabled; 1=enabled. |

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**Table 9: Avaya Agent Deskphone 16CC Customizeable System Parameters (continued)**

| Parameter Name | Default Value | Description and Value Range |
|---|---|---|
| ENABLE_G726 | 0 | Enable or disable G726 capability of the telephone. If the parameter is set to 1, the telephone includes G726 capability in an outbound INVITE request, and accepts G726 when received in an incoming INVITE request. Values are 0=disabled, off; 1=enabled, on. |
| ENABLE_G729 | 1 | Enable or disable G729A codec capability of the phone. Values are:<br>0=G729A disabled.<br>1=The phone includes G729(A) without Annex B support capability in an outbound INVITE request, and accepts G729 when received in an incoming INVITE request.<br>2=The phone includes G729(A) with Annex B support capability in an outbound INVITE request, and accepts G729 when received in an incoming INVITE request. |
| ENABLE_MODIFY_ CONTACTS | 1 | Enable or disable the ability to modify contacts if the Contact application is enabled. Values are 0=disabled; 1=enabled. |
| ENABLE_MULTIPLE_ CONTACTS_WARNING | 1 | Activate/deactivate multiple contacts warning. Depending on current value, a warning message is displayed explaining to the user there are multiple devices registered on user's behalf and that this can cause service disruption. Values: 0 = warning disabled, 1 = warning enabled. |
| ENABLE_REDIAL | 1 | Enable or disable complete Redial functionality. If disabled pressing the redial button has no effect and the redial softkeys and menu items are not displayed. Values are 0=disabled; 1=enabled. |
| ENABLE_REDIAL_LIST | 1 | Enables or disables the capability to redial out of a list of recently dialed numbers instead of performing last number redial. Values are 0=disabled (last number redial only is offered to the user); 1=enabled (user can select either last number redial or redial from a list). |
| ENHDIALSTAT | 1 | Enhanced Dialing Status. Valid range is 0 to 2. If set to “0” the feature is turned off. If set to “1” it is partially enabled (dialing rules do not apply for dialing from Contacts). If set to “2”, the Enhanced Local Dialing feature is fully enabled (dialing rules also apply for dialing from Contacts). |
| FAILED_SESSION_ REMOVAL_TIMER | 30 | Timer to automatically remove a failed call session. Range in seconds is 5 to 999. When an invalid extension is dialed, the endpoint plays the re-order tone and the session line appearance is shown while the re-order tone plays. At this point the user chooses to end the call by pressing the "End" softkey. If user does not end the session, the session line appearance and the re-order tone stay on for the duration of time determined by this parameter. At the end, the session line appearance is removed and the re-order tone stops. |

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**Table 9: Avaya Agent Deskphone 16CC Customizeable System Parameters (continued)**

| Parameter Name | Default Value | Description and Value Range |
|---|---|---|
| G726_PAYLOAD_TYPE | 110 | RTP dynamic payload used for G.726. Range is 96 to 127. |
| GMTOFFSET | 0:00 | Offset used to calculate time from GMT reference time. Default string length positive or negative number of hours and minutes less than 13 hours. |
| GROUP | 0 | Specific user group as tested in configuration files. Valid values are 0 to 999. |
| HEADSYS | 1 | Headset operational mode. One ASCII numeric digit. Valid values are:<br>0 or 2=General Operation, where a disconnect message returns the telephone to an idle state.<br>1 or 3=Call Center Operation, where a disconnect message does not change the state of the telephone. |
| HTTPDIR | " " (Null) | HTTP server directory path. The path name prepended to all file names used in HTTP and HTTPS get operations during initialization. Value: 0-127 ASCII characters, no spaces. Null is a valid value. Leading or trailing slashes are not required. The command syntax is “GET HTTPDIR *myhttpdir*” where “myhttpdir” is your HTTP server path. HTTPDIR is the path for all HTTP operations. |
| HTTPEXCEPTION DOMAINS | " " (Null) | Domains to be excluded for simplified certification enrollment protocol (SCEP). String representing zero or one domains in a URL of 0 to 255 characters in dotted decimal or DNS name format with multiple domains delimited by commas. |
| HTTPPORT | 80 | Destination TCP port used for requests to the HTTP server during initialization. Range is 0 - 65535.<br>**Note:** For this release, there should be no need to set this parameter to values other than default value. |
| HTTPPROXY | " " (Null) | Zero or one IP or DNS addresses of the HTTP server for SCEP. 0 to 255 characters in dotted decimal or DNS name format followed by a colon and port number. The colon and port number are optional. If this parameter is not null, this (proxy) transport address is used to set up the HTTP connection as the transport protocol for SCEP. |
| HTTPSRVR | 0.0.0.0 | IP Address(es) or DNS Name(s) of HTTP file server(s) used to download telephone files (settings file, language files, code). HTTP server addresses can be in dotted decimal or DNS format, and must be separated by commas (0-255 ASCII characters, including commas). |
| ICMPDU | 1 | Controls whether ICMP Destination Unreachable messages will be processed. Values are:<br>0=DU messages not transmitted.<br>1= DU messages not transmitted in response to specific events.<br>2= DU message with code 2 will be transmitted in case of specific events. |

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**Table 9: Avaya Agent Deskphone 16CC Customizeable System Parameters (continued)**

| Parameter Name | Default Value | Description and Value Range |
|---|---|---|
| ICMPRED | 0 | Controls whether ICMP Redirect messages will be processed. Values are:<br>0 = Redirect messages will neither be transmitted nor received Redirect messages will be supported.<br>1 = Redirect messages will not be transmitted, but received Redirect messages will be supported per RFC 1122. |
| INTER_DIGIT_TIMEOUT | 5 | This is the timeout that takes place when user stops inputting digits. The timeout is treated as digit collection completion, and when it occurs, the application sends out an invite. Range in seconds of 1 to 10. |
| IPADD | 0.0.0.0 | IP Address of the telephone. Range is 7 to 15 ASCII characters (less than the default string length) defining one IP Address in dotted-decimal format. |
| L2Q | 0 | Requests 802.1Q tagging mode (auto/on/off). Values are:<br>0 = auto<br>1 = on<br>2 = off |
| L2QAUD | 6 | Layer 2 audio priority value. Range from 0 to 7. |
| L2QSIG | 6 | Layer 2 signaling priority value. Range from 0 to 7. |
| L2QVLAN | n/a | 802.1Q VLAN Identifier (0 to 4094). Null (" ") is not a valid value and the value cannot contain spaces. This parameter is preserved in RAM which survives reset and stored to flash (as L2QVLAN_INIT) only upon successful registration. This value is initialized from L2QVLAN_INIT after power-up. This value will not be initialized from L2QVLAN_INIT after reset, but can be modified using the ADDR craft procedure. |
| LANG0STAT | 1 | This flag defines whether or not the built-in English is offered to the user as a selectable item in the language selection UI menu. At least one other language file must be downloaded before "not offering" built-in English. Values are 0=not offered; 1=selectable. |

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**Table 9: Avaya Agent Deskphone 16CC Customizeable System Parameters (continued)**

| Parameter Name | Default Value | Description and Value Range |
|---|---|---|
| LANGUAGES | " " (Null) | List of links to language files to be downloaded. Substrings are delimited by commas. Maximum length is 1023 characters. Each substring must follow one of the these naming rules:<br>Substring is identical to a file name without any prefix specifying the path or server. The files are downloaded from the same source as the settings file(s).<br>Substring may provide a prefix to the file name, which specifies the relative path ("../" for next higher directory level) from the directory to which the settings file(s) has been downloaded to the directory the language file will be downloaded.<br>Substring specifies the completed URL to the language file including protocol identifier ("http://" or "https://"), server and path. |
| LOCAL_LOG_LEVEL | 3 | Numerical code of severity level. Store entries to the local event log, if event occurs with a severity level whose numerical code is equal to or less than the LOCAL_LOG_LEVEL value. Values are: 0 (emergencies), 1 (alerts), 2 (critical), 3 (errors), 4 (warning), 5 (notice), 6 (informational), 7 (debug). |
| LOG_CATEGORY | | Comma-separated list of keywords in standard string format representing logging categories (SW modules or functions to be included in lower level logging). Logging implementation blocks all traces at level 4 "Warning" or lower, unless the category corresponding to a given trace is enabled. If the LOCAL_LOG_LEVEL is set to "Warning" or lower, this parameter would enable low-level traces from the adaptors or manager as indicated. Applies to all logging mechanisms (syslog and local log). Example: "ALSIP, SESSION" enables debug level traces from the ALSIP adaptor and Session manager. |
| LOGSRVR | " " (Null) | Syslog server IP or DNS address. 0 to 255 characters: zero or one IP Addresses in dotted decimal or DNS name format. |

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**Table 9: Avaya Agent Deskphone 16CC Customizeable System Parameters (continued)**

| Parameter Name | Default Value | Description and Value Range |
|---|---|---|
| MEDIAENCRYPTION | 9 | This parameter sets the cryptosuite and session parameters for SRTP. The parameter can have one or two of the following nine values (separated by commas without any intervening spaces):<br>1=aescm128-hmac80<br>2=aescm128-hmac32<br>3=aescm128-hmac80-unauth<br>4=aescm128-hmac32-unauth<br>5=aescm128-hmac80-unenc<br>6=aescm128-hmac32-unenc<br>7=aescm128-hmac80-unenc-unauth<br>8=aescm128-hmac32-unenc-unauth<br>9=none |
| MSGNUM | " " (Null) | Voice mail system telephone/extension number. Specifies the number to be dialed automatically when the telephone user presses the **Message** button. |
| MTU_SIZE | 1500 | Maximum Transmission Unit size. Range is 1496 or 1500 only octets. |
| MYCERTCAID | CAIdentifier | Certificate Authority Identifier. String identifying whether the endpoints can work with another certificate authority. |
| MYCERTCN | $SERIALNO | Common name (CN) for SUBJECT in SCEP certificate request. Values are:<br>$SERIALNO = the phone's serial number is included as CN parameter in the SUBJECT of a certificate request.<br>$MACADDR = the phone's MAC address is included as CN parameter in the SUBJECT in the certificate request. |
| MYCERTDN | " " (Null) | Common part of SUBJECT in SCEP certificate request. String which defines the part of SUBJECT in a certificate request (including Organizational Unit, Organization, Location, State, Country), of 0 to 255 characters, starting with / and separating items with /. |
| MYCERTKEYLEN | 1024 | Private Key length in range of 1024 to 2048. |
| MYCERTRENEW | 90 | Threshold to renew certificate (given as percentage of device certificate's Validity Object). Range is 1 to 99. |
| MYCERTURL | " " (Null) | URL of SCEP server. String representing zero or one URI starting with "http://", 0 to 255 characters. |
| MYCERTWAIT | 1 | Flag defining phone's behavior when performing certificate enrollment. Values are:<br>0=wait until a certificate or a denial is received or a pending notification is received.<br>1=periodical check in the background. |
| NETMASK | 0.0.0.0 | IP subnet mask. Range is 7 to 15 ASCII characters defining one IP Address in dotted-decimal format. |

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**Table 9: Avaya Agent Deskphone 16CC Customizeable System Parameters (continued)**

| Parameter Name | Default Value | Description and Value Range |
|---|---|---|
| NO_DIGITS_TIMEOUT | 20 | Number of seconds of delay after going "off-hook" or getting secondary dial tone before phone automatically plays a warning tone and does not accept dial input any longer. Range in seconds is 1 to 60. |
| OUTBOUND_ SUBSCRIPTION_ REQUEST_DURATION | 86400 | Number of seconds used in initial SUBSCRIBE messages. This is the suggested duration value of the telephone, which might be lowered by the server, depending on the server configuration. Range is 60-31536000. Note that the default value is equal to one day and the maximum value represents one year. |
| PHNCC | 1 | Telephone country code. The administered international country code for the location by the algorithm that dials calls from the incoming Call Log. Range: 1-3 digits, from “1” to “999.” |
| PHNDPLENGTH | 5 | Internal extension telephone number length. Specifies the number of digits associated with internal extension numbers by the algorithm that dials calls from the incoming Call Log. Range: 1 or 2 digits, from “3” to “13.” |
| PHNIC | 011 | Telephone international access code. The maximum number of digits, if any, dialed to access public network international trunks by the algorithm that dials calls from the incoming Call Log. Range: 0-4 digits. |
| PHNLD | 1 | Telephone long distance access code. The digit, if any, dialed to access public network long distance trunks. Range: 1 digit (0 to 9) or " " (Null). Needed for "Enhanced Local Dialing Algorithm." |
| PHNLDLENGTH | 10 | Length of national telephone number. The number of digits in the longest possible national telephone number. Range: 5 to 15. Needed to for "Enhanced Local Dialing Algorithm." |
| PHNOL | 9 | Outside line access code. The character(s) dialed, including # and *, if any, to access public network local trunks. Range: 0-2 dialable numeric digits, including " " (Null). |
| PHY1STAT | 1 | Ethernet line interface setting (1=auto-negotiate, 2=10Mbps half-duplex, 3=10Mbps full-duplex, 4=100Mbps half-duplex, 5=100Mbps full-duplex, and 6=1000Mbps full-duplex if supported by the hardware). |
| PHY2PRIO | 0 | Layer 2 priority value for frames received on or forwarded to the secondary Ethernet interface. Set this parameter only when VLAN separation is "1" (enabled). Values are from 0-7 and correspond to the drop-down menu selection. |

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**Table 9: Avaya Agent Deskphone 16CC Customizeable System Parameters (continued)**

| Parameter Name | Default Value | Description and Value Range |
|---|---|---|
| PHY2STAT | 1 | Secondary Ethernet interface setting (0=Secondary Ethernet interface off/disabled, 1=auto-negotiate, 2=10Mbps half-duplex, 3=10Mbps full-duplex, 4=100Mbps half-duplex, 5=100Mbps full-duplex), and, for post-Release S1.0 use, 6=1000Mbps full-duplex (if supported by the hardware). |
| PHY2VLAN | 0 | VLAN identifier used by frames received on or forwarded to the secondary Ethernet interface. Set this parameter only when VLAN separation is “1” (enabled). Value is 1-4 ASCII numeric digits from “0” to “4094.” Null is not a valid value, nor can the value contain spaces. |
| PROCPSWD | 27238 | Text string containing the local (dialpad) procedure password (Null or 1-7 ASCII digits). If set, password must be entered immediately after accessing the Craft Access Code Entry screen, either during initialization or when Mute is pressed to access a craft procedure. Intended to facilitate restricted access to local procedures even when command sequences are known. Password is viewable, not hidden. |
| PROCSTAT | 0 | Controls access to local (dialpad) administrative procedures. Values are:<br>0 = Full access to craft local procedures.<br>1 = restricted access to craft local procedures. |
| PROVIDE_ LOGOUT | 1 | Flag to define whether or not logout function is provided to user. If disabled and phone is operating in user mode, hide "Logout" item in A (Avaya) menu. Values are: 0=off; 1=on. |
| PROVIDE_ NETWORKINFO_ SCREEN | 1 | Flag to define whether or not "Network Information" menu is provided to user. If disabled and phone is operating in user mode, hide complete "Network Information" option/sub-menu. Values are: 0=off; 1=on. |
| PROVIDE_OPTIONS_ SCREEN | 1 | Flag to define whether or not "Options & Settings" menu is provided to user. If disabled and phone is operating in user mode, hide complete "Option & Settings" menu tree. Values are: 0=off; 1=on. |
| REGISTERWAIT | 3600 | Number of seconds for next re-registration to SIP server. Range in seconds 10 - 1 000 000 000. |
| ROUTER | 0.0.0.0 | Address(es) of default router(s) / gateway(s) in the IP network. Range is 7-127 characters defining one or more IP Addresses in dotted decimal format, separated by commas without any intervening spaces. |
| RTCPCONT | 1 | Enables/disables the RTCP in parallel to RTP audio streams. Values are 0=RTCP disabled, 1=RTCP enabled. |
| RTP_PORT_LOW | 5004 | Specifies lower limit of a port range to be used by RTP/ RTCP or SRTP/SRTCP connections, for example, to adapt to firewall traversal policies. Values: 1024-65503. |

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**Table 9: Avaya Agent Deskphone 16CC Customizeable System Parameters (continued)**

| Parameter Name | Default Value | Description and Value Range |
|---|---|---|
| RTP_PORT_RANGE | 40 | Specifies the width of the port range to be used by RTP/RTCP or SRTP/SRTCP connections, for example, to adapt to firewall traversal policies. The upper limit is calculated by the value of RTP_PORT_LOW plus the value of RTP_PORT_RANGE, taking into consideration the overall limit of 65535. Values: 32-64511. |
| SEND_DTMF_ TYPE | 2 | Defines whether DTMF tones are send in-band (regular audio) or out-band (negotiation and transmission of DTMF according to RFC 2833, with fallback to send in-band DTMF tones, if far end does not support RFC2833). Values are 1=in-band DTMF; 2=RFC2833 procedure. |
| SIG | 0 | Parameter to allow to download during start-up the specific configuration sets for H323 or SIP endpoints. Valid values are:<br>0=Default<br>1=H323<br>2=SIP |
| SIG_PORT_LOW | 1024 | Lower limit of port range for signaling to support by the phone. Values range from 1024 to 65503. |
| SIG_PORT_RANGE | 64511 | Port range for signaling to support by the phone. Values range from 32 to 64511. |
| SIP_PORT_SECURE | 5061 | Default SIP port (for secure message transfer via TLS). Values range from 1024 - 65535. |
| SIPDOMAIN | " " (Null) | SIP domain name for registration. 0 to 255 characters: string representing domain name. |
| SIPPORT | 5060 | Default SIP port (for non-secure message transfer only). Values range from 1024 - 65535. |
| SIPROXYSRVR | " " (Null) | SIP proxy/registrar server IP or DNS address. Zero or one IP Address. Format is dotted decimal or DNS format, separated by commas, with no spaces (0-255 ASCII characters, including commas). |
| SIPSIGNAL | 2 | SIP signaling transport protocol. Values are:<br>0=UDP<br>1=TCP<br>2=TLS over TCP |
| SNMPADD | " " (Null) | Text string containing zero or more allowable source IP Addresses for SNMP queries, in dotted decimal or DNS format, separated by commas, with up to 255 total ASCII characters including commas and no intervening spaces. |
| SNMPSTRING | " " (Null) | Text string containing the SNMP community name string (up to 32 ASCII characters, no spaces). |

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**Table 9: Avaya Agent Deskphone 16CC Customizeable System Parameters (continued)**

| Parameter Name | Default Value | Description and Value Range |
|---|---|---|
| SNTPSRVR | " " (Null) | Used to retrieve date and time via SNTP (in case of several entries the first address is always first, etc.). Zero to 255 characters: zero or more IP Addresses in dotted decimal or DNS name format, separated by commas without any intervening spaces. |
| SYSTEM_ LANGUAGE | " " (Null) | System Default Language definition. String representing a file name (shall be identical to one of the file names received via LANGUAGES parameter or null). |
| TCP_KEEP_ALIVE_ INTERVAL | 10 | Time interval (number of seconds) after which TCP keep-alive packets are re-transmitted. The interval is started by the system TCP/IP stack (when TCP keep-alive is enabled with specified time intervals). Values are 5-60 seconds. |
| TCP_KEEP_ALIVE_ STATUS | 1 | Indicates whether TCP/IP keep-alive should be enabled at the system. Values are 0=TCP keep alive disabled, 1=TCP keep alive enabled. |
| TCP_KEEP_ALIVE_TIME | 60 | This time interval is the time Avaya Agent Deskphone 16CC telephones will wait before sending out a TCP keep-alive message (TCP ACK message) to the far-end. The time is controlled by the system's TCP/IP stack. The timer is restarted after application level data (for example, a SIP message) is sent over the socket. When the system is idle, this keep-alive time expires and results in sending a TCP ACK (keep-alive) packet. Valid values are 10-3600 (seconds). |
| TIMEFORMAT | 0 | Display time according to defined format in the top line and in the call log. Values are:<br>0=am/pm format<br>1=24h format |
| TLSDIR | " " (Null) | Path name for https downloads. Character string of 0 to 127 characters representing a directory name or path to directory. |
| TLSPORT | 443 | Destination TCP port used for requests to https server during initialization. Values: 0-65535. |
| TRUSTCERTS | " " (Null) | File names of certificates to be used for authentication. List of file names separated by commas (0 to 1024 characters). |
| USE_QUAD_ZEROS_ FOR_HOLD | 0 | Flag that indicates whether a= directional attributes or 0.0.0.0 IP Address is used in the SDP to signal hold operation. 0=use “a= directional attributes”, 1=use quad zeros. |

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**Table 9: Avaya Agent Deskphone 16CC Customizeable System Parameters (continued)**

| Parameter Name | Default Value | Description and Value Range |
|---|---|---|
| VLANSEP | 1 | Enables or disables VLAN separation. Controls whether frames to/from the secondary Ethernet interface receive IEEE 802.1Q tagging treatment. The tagging treatment enables frames to be forwarded based on their tags in a manner separate from telephone frames. If tags are not changed, no tag-based forwarding is employed. Values are: 1=On/Enabled, 2= Off/Disabled. This parameter is used with several related parameters. For more information, see VLAN Separation on page 84. |
| VLANTEST | 60 | Number of seconds to wait for a DHCPOFFER when using a non-zero VLAN ID (1-3 ASCII digits, from “0” to “999”). |
| WAIT_FOR_ REGISTRATION_TIMER | 10 | Time in seconds the SIP application will wait for a register response message. If no message is received, registration is retried. Range is 1-60 (seconds). |

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**Note:**

Table 9 applies to all Avaya Agent Deskphone 16CC telephones, however certain 16CC telephones might have additional, optional information that you can administer. For more information, see Chapter 9: Administering Applications and Options.

# VLAN Considerations

This section contains information on how to administer 16CC telephones to minimize registration time and maximize performance in a Virtual LAN (VLAN) environment. If your LAN environment does not include VLANs, set the system parameter L2Q to 2 (off) to ensure correct operation.

## VLAN Tagging

IEEE 802.1Q tagging (VLAN) is a useful method of managing VoIP traffic in your LAN. Avaya recommends that you establish a *voice* VLAN, set L2QVLAN to that VLAN, and provide voice traffic with priority over other traffic. You can set VLAN tagging manually, by DHCP, or in the 46xxsettings.txt file.

If VLAN tagging is enabled (L2Q= 0 or 1), the Avaya Agent Deskphone 16CC telephones set the VLAN ID to L2QVLAN, and the VLAN priority for packets from the telephone to L2QAUD for audio packets and L2QSIG for signalling packets. The default value (6) for these parameters is the recommended value for voice traffic in IEEE 802.1D.

Regardless of the tagging setting, an Avaya Agent Deskphone 16CC telephone will always transmit packets from the telephone at absolute priority over packets from secondary Ethernet. The priority settings are useful only if the downstream equipment is administered to give the *voice* VLAN priority.

## VLAN Detection

The telephones support automatic detection of the condition where the L2QVLAN setting is incorrect. When VLAN tagging is enabled (L2Q= 0 or 1) initially the telephone transmits DHCP messages with IEEE 802.1Q tagging and the VLAN set to L2QVLAN. The telephones will continue to do this for VLANTEST seconds.

- If the VLANTEST timer expires and L2Q=1, the telephone sets L2QVLAN=0 and transmits DHCP messages with the default VLAN (0).
- If the VLANTEST timer expires and L2Q=0, the telephone sets L2QVLAN=0 and transmits DHCP messages without tagging.
- If VLANTEST is 0, the timer will never expire.

**Note:**

Regardless of the setting of L2Q, VLANTEST, or L2QVLAN, you must have DHCP administered so that the telephone will get a response to a DHCPDISCOVER when it makes that request on the default (0) VLAN.

After VLANTEST expires, if a telephone receives a non-zero L2QVLAN value, the telephone will release the IP Address and send DHCPDISCOVER on that VLAN. Any other release will require a manual reset before the telephone will attempt to use a VLAN on which VLANTEST has expired. See the Reset procedure in Chapter 3 of the *Avaya Agent Deskphone 16CC Installation and Maintenance Guide*.

The telephone ignores any VLAN ID administered on the Communication Manager call server.

## VLAN Default Value and Priority Tagging

The system value **L2QVLAN** is initially set to “0” and identifies the 802.1Q VLAN Identifier. This default value indicates “priority tagging” as defined in IEEE 802.IQ Section 9.3.2.3. Priority tagging specifies that your network closet Ethernet switch automatically insert the switch port default VLAN without changing the user priority of the frame (cf. IEEE 802.1D and 802.1Q).

The VLAN ID = 0 (zero) is used to associate priority-tagged frames to the port/native VLAN of the ingress port of the switch. But some switches do not understand a VLAN ID of zero and require frames tagged with a non-zero VLAN ID.

If you do not want the default VLAN to be used for voice traffic:

- Ensure that the switch configuration lets frames tagged by the 16CC telephone through without overwriting or removing them.
- Set the system value **L2QVLAN** to the ***VLAN ID*** appropriate for your voice LAN.

Another system value you can administer is **VLANTEST**. VLANTEST defines the number of seconds the telephone waits for a DHCPOFFER message when using a non-zero VLAN ID. The VLANTEST default is “60” seconds. Using VLANTEST ensures that the telephone returns to the default VLAN if an invalid VLAN ID is administered or if the phone moves to a port where the L2QVLAN value is invalid. The default value is long, allowing for the scenario that a major power interruption is causing the phones to restart. Always allow time for network routers, the DHCP servers, etc. to be returned to service. If the telephone restarts for any reason and the VLANTEST time limit expires, the telephone assumes the administered VLAN ID is invalid. The telephone then initiates registration with the default VLAN ID.

Setting **VLANTEST** to “**0**” has the special meaning of telling the phone to use a non-zero VLAN indefinitely to attempt DHCP. In other words, the telephone does not return to the default VLAN.

**Note:**

If the telephone returns to the default VLAN but must be put back on the L2QVLAN VLAN ID, you must Reset the telephone. See the Reset procedure in the *Avaya Agent Deskphone 16CC Installation and Maintenance Guide*.

## VLAN Separation

VLAN separation is available to control priority tagging from the device on the secondary Ethernet, typically PC data. The following system parameters control VLAN separation:

- **VLANSEP** - enables (1) or disables (0) VLAN separation.
- **PHY2VLAN** - provides the VLAN ID for tagged frames received on the secondary Ethernet interface.
- **PHY2PRIO** - the layer 2 priority value to be used for tagged frames received on the secondary Ethernet interface.

Table 10 provides several VLAN separation guidelines.

**Table 10: VLAN Separation Rules**

| If | | Then |
|---|---|---|
| VLANSEP is “0” | **OR** the telephone is not tagging frames,<br><br>**OR** the telephone is tagging frames with a VLAN ID equal to PHY2VLAN. | Frames received on the secondary Ethernet interface will not be changed before forwarding. For example, tagging is not added or removed and the VLAN ID and tagged frames priority are not changed. The Ethernet switch forwarding logic determines that frames received on the Ethernet line interface are forwarded to the secondary Ethernet interface or to the telephone without regard to specific VLAN IDs or the existence of tags. |
| VLANSEP is “1” (On/Enabled) | | All tagged frames received on the secondary Ethernet interface are changed before forwarding to make the VLAN ID equal to the PHY2VLAN value and the priority value equal to the PHY2PRIO value.<br>Untagged frames received on the secondary Ethernet interface are not changed before forwarding.<br>Tagged frames with a VLAN ID of zero (priority-tagged frames) will either be:<br>- forwarded without being changed (preferred), or<br>- changed before they are forwarded such that the VLAN ID of the forwarded frame is equal to the PHY2VLAN value and the priority value is equal to the PHY2PRIO value. |
| VLANSEP is “1” (On/Enabled) | **AND** the telephone is not tagging frames,<br><br>**OR** if the telephone is tagging frames with a VLAN ID equal to PHY2VLAN,<br><br>**OR** if the PHY2VLAN value is zero. | The Ethernet switch forwarding logic determines that frames received on the Ethernet line interface are forwarded to the secondary Ethernet interface or to the telephone without regard to specific VLAN IDs or the existence of tags.<br><br>Frames received on the secondary Ethernet interface will not be changed before forwarding. In other words, tagging is not added or removed, and the VLAN ID and priority of tagged frames is not changed. |
| VLANSEP is “1” (On/Enabled) | **AND** the telephone is tagging frames with a VLAN ID not equal to PHY2VLAN,<br><br>**AND** the PHY2VLAN value is not zero. | Tagged frames received on the Ethernet line interface will only be forwarded to the secondary Ethernet interface if the VLAN ID equals PHY2VLAN.<br>Tagged frames received on the Ethernet line interface will only be forwarded to the telephone if the VLAN ID equals the VLAN ID used by the telephone.<br>Untagged frames will continue to be forwarded or not forwarded as determined by the Ethernet switch forwarding logic.<br>Tagged frames with a VLAN ID of zero (priority-tagged frames) will either be:<br>- forwarded to the secondary Ethernet interface or the telephone as determined by the forwarding logic of the Ethernet switch (preferred), or<br>- dropped. |

# DNS Addressing

The 16CC supports DNS addresses and dotted decimal addresses. The telephone attempts to resolve a non-ASCII-encoded dotted decimal IP Address by checking the contents of DHCP Option 6. See DHCP Generic Setup on page 52 for information. At least one address in Option 6 must be a valid, non-zero, dotted decimal address, otherwise, DNS fails. The text string for the **DOMAIN** system parameter (Option 15, Table 9) is appended to the address(es) in Option 6 before the telephone attempts DNS address resolution. If Option 6 contains a list of DNS addresses, those addresses are queried in the order given if no response is received from previous addresses on the list. As an alternative to administering DNS by DHCP, you can specify the DNS server and/or Domain name in the HTTP script file. But first **SET** the **DNSSRVR** and **DOMAIN** values so you can use those names later in the script.

**Note:**

Administer Options 6 and 15 appropriately with DNS servers and Domain names respectively.

# Local Administrative (Craft) Options Using the Telephone Dialpad

The *Avaya Agent Deskphone 16CC Installation and Maintenance Guide* details how to use Craft local procedures at the telephone for administration. The local procedures you might use most often as an administrator are:

- **ADDR** - Static address programming.
- **CLEAR** - Remove all administered values, user-specified data, option settings, etc. and return a telephone to its initial “out of the box” default values.
- **DEBUG** - Enable or disable debug mode for the button module serial port.
- **GROUP** - Set the group identifier on a per-phone basis.
- **INT** - Locally enable or disable the secondary Ethernet hub.
- **RESET** - Reset the telephone to default values including the registration extension and password, any values administered through local procedures, and values previously downloaded using DHCP or a settings file.
- **RESTART** - Restart the telephone in response to an error condition, including the option to reset system values.
- **SIP** - Configure SIP call settings.
- **TEST** - Tests the lamps on the telephone.
- **VIEW** - Review the system parameters for the telephone to verify current values and file versions.

**Note:**

Avaya Agent Deskphone 16CC telephones are factory-set with the SIP protocol. Therefore, you do not have to use the SIG local procedure to set SIP as the telephone’s default protocol.

# Language Selection

Avaya Agent Deskphone 16CC telephones are factory-set to display information in the English language. In addition to English, SIP software downloads include the following language files:

- Canadian French
- Parisian French
- Latin American Spanish
- German
- Brazilian Portuguese

Administrators can specify from one to four languages per telephone to replace English. End users can then select which of those languages they want their telephone to display.

All downloadable language files contain all the information needed for the telephone to present the language as part of the user interface.

Use the configuration file (46xxsettings.txt) and these parameters to customize the settings for up to four languages:

- **LANGUAGES** - the list of languages to be downloaded from which the end user can select a desired display language. Each language is listed in the following format: Mlf_{language_name}.xml, for example, Mlf_German.xml.*
- **SYSTEM_LANGUAGE** - a string indicating the filename of the default system language. The string indicates which of the available languages to use for display purposes. If this parameter is not set, or if no other language has been set by the user, or if a user language choice cannot be satisfied, the built-in English strings are used.
- **LANG0STAT** - Allows the user to select the built-in English language when other languages are downloaded. If LANG0STAT is "0" and at least one language is downloaded, the user cannot select the built-in English language. If LANG0STAT is "1" (the default) the user can select the built-in English language text strings.

For more information, see Avaya Agent Deskphone 16CC Customizeable System Parameters.

**Note:**

Specifying a language other than English in the configuration file has no impact on Avaya Communication Manager settings, values, or text strings.

# Enhanced Local Dialing

The 16CC telephones have a variety of telephony-related applications that might obtain a telephone number during operation. For example, the Call Log saves a number of an incoming caller, but does not consider that the user has to then prepend the saved number with a digit to dial an outside line, and possibly a digit to dial long distance.

Avaya Agent Deskphone 16CC telephones can evaluate a raw telephone number, based on administered parameters. The telephone can automatically prepend the correct digits, saving the user time and effort. This is the Enhanced Local Dialing feature. The key to the success of this feature is accurate administration of several important values, summarized below.

The system values relevant to the Enhanced Dialing Feature are:

- **ENHDIALSTAT** - Enhanced dialing status. If set to “1” the enhanced local dialing feature is partially enabled, meaning dialing rules do not apply to dialing from the Contacts list. If set to “2” the enhanced local dialing feature is fully enabled and does apply to dialing from the Contacts list. If set to “0” enhanced local dialing is off.
- **PHNCC** - the international country code of the Communication Manager (CM) call server. For example, “1” for the United States, “44” for the United Kingdom, and so on.
- **PHNDPLENGTH** - the length of the dial plan on the CM call server.
- **PHNIC** - the digits the CM call server dials to access public network international trunks. For example, “011” for the United States.
- **PHNLD** - the digit dialed to access public network long distance trunks on the CM call server.
- **PHNLDLENGTH** - the maximum length, in digits, of the national telephone number for the country in which the CM call server is located.
- **PHNOL** - the character(s) dialed to access public network local trunks on the CM call server.

**Note:**

In all cases, the values you administer are the values relevant to the location of the CM call server at which a 16CC telephone is registered. If a telephone is in Japan, but its CM call server is in the United States, set the **PHNCC** value to “1" for the United States.

In all cases, the digits the telephones insert and dial are subject to standard CM call server features and administration. This includes Class of Service (COS), Class of Restriction (COR), Automatic Route Selection (ARS), and so on.

As indicated in Table 9, you can administer the system parameter **ENHDIALSTAT** to turn off the Enhanced Local Dialing feature.

**Example:** A user initiates an international call. The telephone determines the number to be called is from another country code. The telephone then prepends the rest of the telephone number with PHNOL to get an outside line and PHNIC to get an international trunk. The telephone then dials normally, with the CM call server routing the call appropriately.

# Enhanced Local Dialing Requirements

The enhanced local dialing feature is invoked when all the following conditions are met:

- An application on an Avaya Agent Deskphone 16CC obtains or otherwise identifies a character string as containing a telephone number the user wants to dial, and
- The Phone application determines a call appearance is available for an outgoing call, and
- The originating application passes the character string to the Phone application, and
- The originating application specifies a Source Flag set to **No**, and
- The current value of ENHDIALSTAT is “1” (partially enabled) or “2” (fully enabled).

The Phone application takes the incoming character string, applies an algorithm, and determines the string of digits to be sent to Avaya Communication Manager (CM) for dialing. At this point the Phone application goes off-hook and sends the digits to CM.

The Source Flag has two possible values:

- Yes - the Called Party Number has been administered or otherwise identified as a valid outgoing phone number, such as a Speed Dial button, Redial number, or Outgoing Call Log number, or
- No - the Called Party Number comes from a source that is likely to require enhanced local dialing processing, for example, the Incoming Call Log application.

**Note:**

The Enhanced Local Dialing algorithm requires that telephone numbers be presented in a standard format. The standard format depends on how you administer the parameters indicated in Table 9. The algorithm also assumes that international telephone numbers are identified as such in, for example, the Contacts application. Precede international numbers with a plus (+) sign, and a space or some non-digit character following the country code.

# Agent Confirmation Tones

The telephone provides agent confirmation tones, controlled by the ID number in the AGENTTONE configuration parameter. The tones, by ID number, are the following:

| ID # | Agent Tone | Frequency (Hz) | Cadence (seconds) |
|---|---|---|---|
| 1 | Agent Confirmation (no agent message waiting) | 350+440 | 3x(0.15on, 0.15off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 2 | Agent Confirmation (no agent message waiting) | 425 | 3x(0.15on, 0.15off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 3 | Agent Confirmation (no agent message waiting) | 404 | 3x(0.1on, 0.1off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 4 | Agent Confirmation (no agent message waiting) | 425 | 3x(0.1on, 0.1off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 5 | Agent Confirmation (no agent message waiting) | 425 | 19x(0.15on, 0.15off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 6 | Agent Confirmation (no agent message waiting) | 425 | 3x(0.15on, 0.15off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 7 | Agent Confirmation (no agent message waiting) | 425 | 3x(0.1on, 0.01off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 8 | Agent Confirmation (no agent message waiting) | 425 | 19x(0.05on, 0.05off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 9 | Agent Confirmation (no agent message waiting) | 425 | 3x(0.15on, 0.15off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 10 | Agent Confirmation (no agent message waiting) | 350+440 | 3x(0.1on, 0.1off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 11 | Agent Confirmation (no agent message waiting) | 425 | 19x(0.05on, 0.05off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |

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| ID # | Agent Tone | Frequency (Hz) | Cadence (seconds) |
|---|---|---|---|
| 12 | Agent Confirmation (no agent message waiting) | 330 | 3x(0.1on, 0.1off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 13 | Agent Confirmation (no agent message waiting) | 425 | 19x(0.05on, 0.05off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 14 | Agent Confirmation (no agent message waiting) | 425 | 19x(0.05on, 0.05off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 15 | Agent Confirmation (no agent message waiting) | 425 | 19x(0.05on, 0.05off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 16 | Agent Confirmation (no agent message waiting) | 350+440 | 3x(0.15on, 0.15off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 17 | Agent Confirmation (no agent message waiting) | 350+440 | 3x(0.15on, 0.15off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 18 | Agent Confirmation (no agent message waiting) | 330 | 3x(0.1on, 0.1off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 19 | Agent Confirmation (no agent message waiting) | 350+440 | 3x(0.15on, 0.15off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 20 | Agent Confirmation (no agent message waiting) | 350+440 | 3x(0.15on, 0.15off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 21 | Agent Confirmation (no agent message waiting) | 350+440 | 3x(0.15on, 0.15off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 22 | Agent Confirmation (no agent message waiting) | 350+440 | 3x(0.15on, 0.15off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 23 | Agent Confirmation (no agent message waiting) | 350+440 | 3x(0.15on, 0.15off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 24 | Agent Confirmation (no agent message waiting) | 425 | 3x(0.1on, 0.1off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 25 | Agent Confirmation (no agent message waiting) | 425 | 19x(0.05on, 0.05off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |

| ID # | Agent Tone | Frequency (Hz) | Cadence (seconds) |
|---|---|---|---|
| 26 | Agent Confirmation (no agent message waiting) | 480 | 1 second, then off |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on,0.05off) |
| 27 | Agent Confirmation (no agent message waiting) | 425 | 0.1on, 0.1off, 0.1on,0.1off |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 28 | Agent Confirmation (no agent message waiting) | 400x24 | 0.1on, 0.1off, 0.3on,0.3off |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on,0.05off) |
| 29 | Agent Confirmation (no agent message waiting) | 350 | 0.128on, 0.128off, 0.384on |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 30 | Agent Confirmation (no agent message waiting) | 400 | 0.15on, 0.15off, 0.3on |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 31 | Agent Confirmation (no agent message waiting) | 400 | 20x(0.125on, 0.125off) |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on.0.05off) |
| 32 | Agent Confirmation (no agent message waiting) | 950 | 1 second, then off |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on, 0.05off) |
| 33 | Agent Confirmation (no agent message waiting) | 450 | 0.04on, 0.04off |
| | Agent Confirmation (agent message waiting) | 440 | 5x(0.05on,0.05off) |

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# Chapter 9: Administering Applications and Options

## Customizing Telephone Applications and Options

This chapter covers configuration options for activating/deactivating options and applications. The Avaya Agent Deskphone 16CC telephones offer the user numerous applications like Contacts, Call Log, Redial, and so on. Each of these applications allows the user to add, delete, or in some cases, edit entries. As the administrator, you might not want the user to have that level of functionality.

In 4600 and 9600 Series H.323 IP Telephones, the parameters APPSTAT (meaning Application permission status) and OPSTAT (meaning Options permission status) control application access and functionality. However, 16CC telephones have a more granular way of assigning functionality, with a specific parameter for each permission, as follows:

- **ENABLE_CALL_LOG** - Allows end user access to the list of unanswered and answered calls. If disabled, the Call Log application is not displayed to the user and calls are not logged.
- **ENABLE_REDIAL** - Allows the end user to redial one to three previously called numbers. If disabled, redialing is not available to the end user.
- **ENABLE_REDIAL_LIST** - Allows the end user to select a number to redial from a list. If disabled, only the previously-dialed number can be redialed.
- **ENABLE_CONTACTS** - Allows end user access to a list of numbers and to make calls by selecting a Contact Name/Number. If disabled, the Contacts application is not displayed to the user and a Contact list cannot be set up or maintained.
- **ENABLE_MODIFY_CONTACTS** - If the Contacts application is enabled (ENABLE_CONTACTS=1), this option allows or prevents the end user from changing or updating the Contact list.
- **PROVIDE_OPTIONS_SCREEN** - If disabled, the Options & Settings menu is not displayed on the Avaya menu. The user cannot change any of the features and options associated with the Options & Settings menu.
- **PROVIDE_NETWORKINFO_SCREEN** - If disabled, the Network Information menu is not displayed on the Avaya menu.
- **PROVIDE_LOGOUT** - If disabled, Logout is not displayed to the user as an option on the Avaya menu. Note that this option is never displayed when an agent is logged in as such. The agent must first log out with a Reason Code (if required) before telephone/extension log out can occur.

These parameters have On (1=enabled)/Off (0=disabled) settings, and are described in detail in Table 9: Avaya Agent Deskphone 16CC Customizeable System Parameters.

**Note:**

To facilitate administration of application-related parameters, the 9600 Series (both SIP and H.323) IP Telephones, 4600 Series IP Telephones, and Avaya Agent Deskphone 16CC telephones use the same ***46xxsettings.txt*** file.

# Backup/Restore of User Settings

Many of the user settings (parameters) are backed up to and restored from the Personal Profile Manager (PPM) within SIP Enablement Services (SES). The following table shows the user parameters and the PPM methods used for the parameter. The “getPPM” method restores the parameter from the PPM to the phone. The “setPPM” method stores the parameter from the phone to the PPM.

| Parameter | Set/Get PPM Method & Parameter |
|---|---|
| AGCHEAD | DeviceData; AgcForHeadset |
| AUDIOPATH | DeviceData; DefaultAudioPath |
| CALL_LOG_ACTIVE | DeviceData; CallHistoryActivated |
| CALL_LOG_BRIDGED | DeviceData; CallHistoryLoggingBridgedCalls |
| CLICKS | DeviceData; ButtonClicksEnabled |
| CURRENT_LOGO | DeviceData; CurrentLogo |
| ERRORTONE | DeviceData; ErrortoneEnabled |
| LANGUAGE_FILE_IN_USE | DeviceData; LanguageFileInUse |
| PERSONALRING | VolumeSettings; RingerCadence |
| PHNSCRONCALL | DeviceData; ShowPhoneScreenOnCall |
| PHNSCRONALERT | DeviceData; ShowPhoneScreenOnAlert |
| PHNTIMERS | DeviceData; DisplayCallTimers |
| PHNREDIAL | DeviceData;EffectOfRedialButton |
| PHNVISUALALERT | DeviceData; UseVisualAlerting |
| RingerVolume | VolumeSettings; RingerVolume |
| ReceiverVolume | VolumeSettings; ReceiverVolume |
| TIMEFORMAT | DeviceData; TimeFormat |
| USER_LANGUAGE_CHOICE | DeviceData; UserPreferredLanguage |

For Contact list backup/restore, the telephone uses the getContactList method request to retrieve the user’s contact list from PPM. This request is issued during PPM Synchronization as specified in PPM Synchronization. The telephone uses the setContactList command to change the contact list on the PPM when the user changes the contact list on the phone.

# Appendix A: Glossary of Terms

| | |
|---|---|
| **802.1D**<br>**802.1Q** | 802.1Q defines a layer 2 frame structure that supports VLAN identification and a QoS mechanism usually referred to as 802.1D. |
| **802.1X** | Authentication method for a protocol requiring a network device to authenticate with a back-end Authentication Server before gaining network access. Applicable Avaya Agent Deskphone 16CC telephones support IEEE 802.1X for pass-through and for Supplicant operation with the EAP-MD5 authentication method. Not currently applicable to 9600 Series SIP IP Telephones running SIP R1.0 software or Avaya Agent Deskphone 16CC telephones. |
| **ARP** | Address Resolution Protocol, used, for example, to verify that the IP Address provided by the DHCP server is not in use by another IP telephone. |
| **Call Server** | In an Avaya SIP environment, the “call server” is the combination of SIP Enablement Services (SES) and Avaya Communication Manager. |
| **CELP** | Code-excited linear-predictive. Voice compression requiring only 16 kbps of bandwidth. |
| **CLAN** | Control LAN, a type of circuit pack. |
| **CNA** | Converged Network Analyzer, an Avaya product to test and analyze network performance. |
| **DHCP** | Dynamic Host Configuration Protocol, an IETF protocol used to automate IP Address allocation and management. |
| **DiffServ** | Differentiated Services, an IP-based QoS mechanism. |
| **DNS** | Domain Name System, an IETF standard for ASCII strings to represent IP Addresses. The Domain Name System (DNS) is a distributed Internet directory service. DNS is used mostly to translate between domain names and IP Addresses. Avaya Agent Deskphone 16CC telephones can use DNS to resolve names into IP Addresses. In DHCP, TFTP, and HTTP files, DNS names can be used wherever IP Addresses were available as long as a valid DNS server is identified first. |
| **H.323** | A TCP/IP-based protocol for VoIP signaling. An alternative to SIP for VoIP signaling. Avaya Agent Deskphone 16CC telephones do not support this protocol. |
| **HTTP** | Hypertext Transfer Protocol, used to request and transmit pages on the World Wide Web. |
| **HTTPS** | A secure version of HTTP. |
| **IETF** | Internet Engineering Task Force, the organization that produces standards for communications on the internet. |

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| | |
|---|---|
| **LAN** | Local Area Network. |
| **LLDP** | Link Layer Discovery Protocol. All IP telephones with an Ethernet interface support the transmission and reception of LLDP frames on the Ethernet line interface in accordance with IEEE standard 802.1AB. Not applicable to 9600 Series SIP IP Telephones running SIP R1.0 software or Avaya Agent Deskphone 16CC telephones. |
| **MAC** | Media Access Control, ID of an endpoint. |
| **Media Channel Encryption** | Encryption of the audio information exchanged between the IP telephone and the call server or far end telephone. |
| **NAPT** | Network Address Port Translation. |
| **NAT** | Network Address Translation. |
| **OPS** | Outboard Proxy SIP. |
| **PPM** | Personal Profile Manager, part of the SIP Enablement Services (SES) platform. PPM is responsible for maintaining and managing end users' personal information in the system. |
| **Proxy Server** | An intermediary entity that acts as both a server and a client for the purpose of making requests on behalf of other clients. A proxy server primarily plays the role of routing, meaning its job is to ensure that a request is sent to another entity "closer" to the targeted user. Proxies are also useful for enforcing policy, for example, making sure a user is allowed to make a call. A proxy interprets, and if necessary, rewrites specific parts of a request message before forwarding it. |
| **PSTN** | Public Switched Telephone Network, the network used for traditional telephony. |
| **QoS** | Quality of Service, used to refer to several mechanisms intended to improve audio quality over packet-based networks. |
| **RSVP** | Resource ReSerVation Protocol, used by hosts to request resource reservations throughout a network. |
| **RTCP** | RTP Control Protocol, monitors quality of the RTP services and can provide real-time information to users of an RTP service. |
| **RTP** | Real-time Transport Protocol. Provides end-to-end services for real-time data such as voice over IP. |
| **SCEP** | Simple Certificate Enrollment Protocol, used to obtain a digital certificate. |
| **SDP** | Session Description Protocol. A well-defined format for conveying sufficient information to discover and participate in a multi session. |
| **SES** | SIP Enablement Services, the Avaya solution for SIP telephony with Avaya Communication Manager. |

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| | |
|---|---|
| **Signaling Channel Encryption** | Encryption of the signaling protocol exchanged between the IP telephone and the call server. Signaling channel encryption provides additional security to the security provided by channel encryption. |
| **SIP** | Session Initiation Protocol. An alternative to H.323 for VoIP signaling. The only protocol that Avaya Agent Deskphone 16CC telephones support. |
| **SNTP** | Simple Network Time Protocol. An adaptation of the Network Time Protocol used to synchronize computer clocks in the internet. |
| **SRTCP** | Secure Real-time Transport Control Protocol. |
| **SRTP** | Secure Real-time Transport Protocol. |
| **System - specific** | Specific to a particular type of call server, for example, Avaya Communication Manager (CM). “System-specific signaling” refers to messages specific to the signaling protocol used by the system, for example, H.323 and/or CCMS messages used by CM and IP Office. “System-specific procedures” refers to telephone software procedures that are specific to the call server with which the software is intended to be used. |
| **TCP/IP** | Transmission Control Protocol/Internet Protocol, a network-layer protocol used on LANs and internets. |
| **TFTP** | Trivial File Transfer Protocol, used to provide downloading of upgrade scripts and binary files to certain IP telephones. 16CC telephones use HTTP or HTTPS instead of TFTP. |
| **TLS** | Transport Layer Security, an enhancement of Secure Sockets Layer (SSL). TLS is compatible with SSL 3.0 and allows for privacy and data integrity between two communicating applications. |
| **TLV** | Type-Length-Value elements transmitted and received as part of Link Layer Discovery Protocol (LLDP). |
| **UDP** | User Datagram Protocol, a connectionless transport-layer protocol. |
| **URI & URL** | Uniform Resource Identifier and Uniform Resource Locator. Names for the strings used to reference resources on the Internet (for example, HTTP://....). URI is the newer term. |
| **VLAN** | Virtual LAN. |
| **VoIP** | Voice over IP, a class of technology for sending audio data and signaling over LANs. |
| **WML** | Wireless Markup Language, used by Web Browsers to communicate with WML servers. Not applicable to Avaya Agent Deskphone 16CC telephones. |

# Appendix B: Related Documentation

## Avaya Call Center 5.0 Documentation

Avaya Agent Deskphone 16CC telephones are the endpoints of Avaya's call center solution. For an overview of the SIP call center solution, Avaya Call Center 5.0, see *Implementing Avaya Call Center 5.0 with Avaya Agent Deskphone 16CC*, available on the Avaya Support Web site, www.avaya.com/support. That document provides a "road map" to related documentation for each segment of Avaya Call Center 5.0.

## IETF Documents

IETF documents provide standards relevant to IP Telephony and are available for free from the IETF Web site: http://www.ietf.org/rfc.html.

## ITU Documents

Access the ITU Web site for more information about ITU guidelines and documents, available for a fee from the ITU Web site: http://www.itu.int.

## ISO/IEC, ANSI/IEEE Documents

Access the ISO/IEC standards Web site for more information about IP Telephony standards, guidelines, and published documents: http://www.iec.ch.

# Appendix C: Sample Station Forms and Other Screens

Use the sample screens that follow as guidelines for Call Center/telephone setup.

**Figure 4: Station screen**

```
display station 3003 Page                                    1 of 6
                                    STATION
Extension: 3003                     Lock Messages? n             BCC: 0
     Type: 4620SIPCC                  Security Code: *            TN: 1
     Port: S00018                   Coverage Path 1:             COR: 8
     Name: Call_Center,3003         Coverage Path 2:             COS: 5
                                    Hunt-to Station:
STATION OPTIONS
                                         Time of Day Lock Table:
              Loss Group: 19        Personalized Ringing Pattern: 1
                                              Message Lamp Ext: 3001
           Speakerphone: 2-way             Mute Button Enabled? y
        Display Language: english             Expansion Module? n
 Survivable GK Node Name:
          Survivable COR: internal            Media Complex Ext:
   Survivable Trunk Dest? y                         IP SoftPhone? y
                                             IP Video Softphone? n
                                           Customizable Labels? Y
```

**Figure 5: Station screen (page 2)**

```
display station 3003                                   Page   2 of   6
                                    STATION
FEATURE OPTIONS
          LWC Reception: spe     Auto Select Any Idle Appearance? n
         LWC Activation? y                Coverage Msg Retrieval? y
 LWC Log External Calls? n                          Auto Answer: none
           CDR Privacy? n                     Data Restriction? n
  Redirect Notification? y           Idle Appearance Preference? n
Per Button Ring Control? n         Bridged Idle Line Preference? n
  Bridged Call Alerting? n             Restrict Last Appearance? y
 Active Station Ringing: single
                                              EMU Login Allowed? n
       H.320 Conversion? n Per Station CPN - Send Calling Number?
      Service Link Mode: as-needed
        Multimedia Mode: enhanced       Audible Message Waiting? n
   MWI Served User Type:              Display Client Redirection? n
            AUDIX Name:             Select Last Used Appearance? n
                                     Coverage After Forwarding? s
Remote Softphone Emergency Calls:as-on-local Direct IP-IP Audio Connections? y
      Emergency Location Ext: 3001 Always Use? n IP Audio Hairpinning? y
```

**Figure 6: Station screen (page 3)**

```
display station 3003                                    Page   3 of   6
                                   STATION
Conf/Trans on Primary Appearance? n
Bridged Appearance Origination Restriction? n
            Call Appearance Display Format: disp-param-default
IP Phone Group ID:

                              ENHANCED CALL FORWARDING
                                    Forwarded Destination         Active
 Unconditional For Internal Calls To:                                n
                   External Calls To:                                n
          Busy For Internal Calls To:                                n
                   External Calls To:                                n
      No Reply For Internal Calls To:                                n
                   External Calls To:                                n
```

**Figure 7: Station screen (page 4)**

```
display station 3003                                    Page   4 of   6
                                   STATION
 SITE DATA
       Room:                                            Headset? n
       Jack:                                            Speaker? n
      Cable:                                           Mounting: d
      Floor:                                       Cord Length: 0
   Building:                                         Set Color:

ABBREVIATED DIALING
     List1:                    List2:                    List3:

BUTTON ASSIGNMENTS
 1: call-appr                          5: manual-in          Grp:
 2: call-appr                          6: after-call         Grp:
 3: call-appr                          7: work-code
 4: aux-work    RC:    Grp:            8:
```

**Figure 8: Station screen (page 6)**

```
display station 3003                                    Page   6 of   6
                                   STATION
SIP FEATURE OPTIONS
         Type of 3PCC Enabled:Avaya
```

**Note:**

Define the Configuration Set (Change Off-PBX Telephone Configuration-Set screen), e.g., set = 1.

**Figure 9: Configuration Set screen**

```
change off-PBX-telephone configuration-set 1

                            CONFIGURATION SET: 1

             Configuration Set Description: Standard
                      Calling Number Style: network
       CDR for Calls to EC500 Destination? y
                     CDR for Origination: phone-number
               Fast Connect on Origination? n
              Post Connect Dialing Options: dtmf
             Cellular Voice Mail Detection: none
                            Barge-in Tone? n
               Calling Number Verification? y
Call Appearance Selection for Origination: primary-first
                         Confirmed Answer? n
```

**Note:**

Add the agent's station extension to the Off-PBX Station Mapping screen (only the OPS Outboard Proxy SIP mode is supported for the 16CC) as follows.

**Important:**

The Trunk Group set up for SIP has to be assigned here.

**Figure 10: Stations with Off-PBX Telephone Integration screen (page 1)**

```
display off-pbx-telephone station-mapping 3003              Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number      Trunk       Config
 Extension                  Prefix                      Selection   Set
 3003             OPS           -     3003               7          1
```

**Figure 11: Stations with Off-PBX Telephone Integration screen (page 2)**

```
display off-pbx-telephone station-mapping 3003              Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping      Calls       Bridged        Location
 Extension     Limit      Mode         Allowed     Calls
 3003           3       both           all         none           1
```

**Note:**

On page 2 of this screen, set the Call Limit to at least the number of appearance buttons assigned to the phone and set Mapping Mode to “both.” Set Calls Allowed to “all” and Bridged Calls to “none” for the SIP OPS mode.

**Figure 12: Stations with Off-PBX Telephone Integration screen (page 1)**

```
display off-pbx-telephone station-mapping 3003              Page   1 of   2
                 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number      Trunk       Config
 Extension                  Prefix                      Selection   Set
 3003             OPS           -     3003               7          1
```

**Figure 13: Stations with Off-PBX Telephone Integration screen (page 2)**

```
display off-pbx-telephone station-mapping 3003              Page   2 of   2
                 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping      Calls       Bridged        Location
 Extension     Limit      Mode         Allowed     Calls
 3003           3       both           all         none           1
```

**Figure 14: SIP Feature Options page**

```
change station nnnn                                          Page 5 of x
                                    STATION

SIP Feature Options
         Type of 3PCC Enabled: none
```

**Figure 15: Feature-Related System Parameters screen**

```
change system-parameters features                              page 11 of x
                       FEATURE-RELATED SYSTEM PARAMETERS

CALL CENTER SYSTEM PARAMETERS
  EAS
        Expert Agent Selection (EAS) Enabled? n
       Minimum Agent-LoginID Password Length:
         Direct Agent Announcement Extension: _________           Delay: ___
   Message Waiting Lamp Indicates Status For: station
VECTORING
                   Converse First Data Delay: 0       Second Data Delay: 2
              Converse Signaling Tone (msec): 100          Pause (msec): 70_
                    Prompting Timeout (secs): 10
                Interflow-qpos EWT Threshold: 2
   Reverse Star/Pound Digit For Collect Step? n
         Available Agent Adjustments for BSR? n
                           BSR Tie Strategy? 1st_found
  Store VDN Name in Station's Local Call Log? n

SERVICE OBSERVING
             Service Observing: Warning Tone? n      or Conference Tone? n
    Service Observing Allowed with Exclusion? n
            Allow Two Observers in Same Call? n
```

**Figure 16: System Parameters Customer-Options (Optional Features) screen**

```
display system-parameters customer-options                    Page   1 of  x
                              OPTIONAL FEATURES

G3 Version: V12  123456789012                  Software Package: Standard
  Location: 2                               RFA System ID (SID):
  Platform: 2                               RFA Module ID (MID): 123456

                                                          USED
                                 Platform Maximum Ports: 300   174
                                       Maximum Stations: 300   174
                                Maximum XMOBILE Stations: 30    28
                     Maximum Off-PBX Telephones - EC500: 1200  0
                     Maximum Off-PBX Telephones -   OPS: 1200  0
                     Maximum Off-PBX Telephones - SCCAN: 0     0
                     Maximum Off-PBX Telephones - PBFMC: 0     0
                     Maximum Off-PBX Telephones - PVFMC: 0     0

        (NOTE: You must logoff & login to effect the permission changes.)
```

**Figure 17: System Parameters Customer-Options (Optional Features) screen**

```
display system-parameters customer-options                     page 2 of x
                              OPTIONAL FEATURES

IP PORT CAPACITIES                                             USED
                        Maximum Administered H.323 Trunks:
                           Maximum Administered IP Trunks: 100   96
             Maximum Concurrently Registered IP Stations: 10    10
               Maximum Administered Remote Office Trunks: 0     0
  Maximum Concurrently Registered Remote Office Stations: 0     0
                Maximum Concurrently Registered IP eCons: 0     0
                           Maximum Video Capable Stations: 0     0
                     Maximum Video Capable IP Softphones: 0     0
                         Maximum Administered SIP Trunks: 500   25
    Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
     Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                                Maximum TN2501 VAL Boards: 1     0
                      Maximum G250/G350/G700 VAL Sources: 0     0
             Maximum TN2602 Boards with 80 VoIP Channels: 20    12
            Maximum TN2602 Boards with 320 VoIP Channels: 4     3

     Maximum Number of Expanded Meet-me Conference Ports: 0      0

           (NOTE: You must logoff & login to effect the permission changes.)
```

**Figure 18: System Parameters Customer-Options (Optional Features) screen**

```
display system-parameters customer-options                      page 3 of x
                        OPTIONAL FEATURES

     Abbreviated Dialing Enhanced List?            Audible Message Waiting?
         Access Security Gateway (ASG)?                Authorization Codes?
         Analog Trunk Incoming Call ID?                         CAS Branch?
  A/D Grp/Sys List Dialing Start at 01?                           CAS Main?
 Answer Supervision by Call Classifier?                  Change COR by FAC?
                                 ARS?    Computer Telephony Adjunct Links?
                ARS/AAR Partitioning?     Cvg Of Calls Redirected Off-net?
           ARS/AAR Dialing without FAC?                       DCS (Basic)?
           ASAI Link Core Capabilities?                 DCS Call Coverage?
           ASAI Link Plus Capabilities?                DCS with Rerouting?
        Async. Transfer Mode (ATM) PNC?
Async. Transfer Mode (ATM) Trunking?         Digital Loss Plan Modification?
               ATM WAN Spare Processor?                           DS1 MSP?
                                 ATMS?               DS1 Echo Cancellation?
                  Attendant Vectoring?
```

**Figure 19: System Parameters Customer-Options (Optional Features) screen**

```
display system-parameters customer-options                    Page   4 of  x
                              OPTIONAL FEATURES
  Emergency Access to Attendant? y                           IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                    ISDN Feature Plus? y
                  Enhanced EC500? y      ISDN/SIP Network Call Redirection? y
   Enterprise Survivable Server? y
      Enterprise Wide Licensing? y                        ISDN-BRI Trunks? y
             ESS Administration? y                               ISDN-PRI? y
         Extended Cvg/Fwd Admin? y             Local Survivable Processor? y
    External Device Alarm Admin? y                   Malicious Call Trace? y
         Extended Cvg/Fwd Admin? y   Mode Code for Centralized Voice Mail? y
    External Device Alarm Admin? y
 Five Port Networks Max per MCC? y               Multifrequency Signaling? y
              Flexible Billing? y Multimedia Appl. Server Interface(MASI)? y
  Forced Entry of Account Codes? y        Multimedia Call Handling (Basic)?
    Global Call Classification? y     Multimedia Call Handling (Enhanced)? y
            Hospitality (Basic)? y             Multimedia IP SIP Trunking? y
Hospitality (G3V3 Enhancements)? y
                      IP Trunks? y
          IP Attendant Consoles? y

           (NOTE: You must logoff & login to effect the permission changes.)
```

**Figure 20: System Parameters Customer-Options (Optional Features) screen**

```
display system-parameters customer-options                      page 5 of x

                              OPTIONAL FEATURES

               Multinational Locations?            Station and Trunk MSP? n
Multiple Level Precedence and Preemption?    Station as Virtual Extension? n
                     Multiple Locations?
                                         System Management Data Transfer? n

         Personal Station Access (PSA)? y
                     Posted Messages? n              Tenant Partitioning? n
                     PNC Duplication? n       Terminal Trans. Init. (TTI)? y
                Port Network Support? y              Time of Day Routing? y
           Processor and System MSP? n             Uniform Dialing Plan? y
                 Private Networking? y     Usage Allocation Enhancements? y
                 Processor Ethernet? y       TN2501 VAL Maximum Capacity? y

                       Remote Office? n               Wideband Switching? y
       Restrict Call Forward Off Net? y                         Wireless? n
                 Secondary Data Module? y
```

**Figure 21: Call Center Optional Features screen**

```
display system-parameters customer-options                      page 6 of x

                        CALL CENTER OPTIONAL FEATURES

                           Call Center Release:

                               ACD? y    PASTE (Display PBX Data on Phone)? n
                     BCMS (Basic)? y                         Reason Codes? n
                                                Service Level Maximizer? y
      BCMS/VuStats Service Level? n            Service Observing (Basic)? y
                Business Advocate? n    Service Observing (Remote/By FAC)? n
                    Call Work Codes? y             Service Observing (VDNs)?
     DTMF Feedback Signals For VRU? y                              Timed ACW?
                 Dynamic Advocate? n                   Vectoring (Basic)? y
    Expert Agent Selection (EAS)? y                  Vectoring (Prompting)? y
                            EAS-PHD? n             Vectoring (G3V4 Enhanced)?
                 Forced ACD Calls? n    Vectoring (ANI/II-Digits Routing)? n
              Least Occupied Agent?     Vectoring (G3V4 Advanced Routing)? n
       Lookahead Interflow (LAI)?                    Vectoring (CINFO)? n
Multiple Call Handling (On Request)? n   Vectoring (Best Service Routing)? n
    Multiple Call Handling (Forced)? n                Vectoring (Holidays)?
```

**Figure 22: Call Center Optional Features screen**

```
display system-parameters customer-options                    Page 7 of x
                        CALL CENTER OPTIONAL FEATURES

       VDN of Origin Announcement? n                              VuStats? n
           VDN Return Destination? n              VuStats (G3V4 Enhanced)? n




                                  Used
              Logged-In ACD Agents: 500
         Logged-In Advocate Agents: 500
     Logged-In IP Softphone Agents: 500
          Logged-In SIP EAS Agents: 50
```

**Figure 23: QSIG Optional Features screen**

```
display system-parameters customer-options                     Page 8 of x
                            QSIG OPTIONAL FEATURES

                                        Basic Call Setup? n
                             Basic Supplementary Services? n
                                    Centralized Attendant? n
                                    Interworking with DCS? n
                      Supplementary Services with Rerouting? n
                                    Transfer into QSIG Voice Mail? n
                                              Value-Added (VALU)? n
```

**Figure 24: ASAI Features screen when the ASAI Link Plus Capabilities field is y**

```
change system-parameters customer options                      Page 8 of X

                                    ASAI FEATURES

                                   CTI Stations? n
                                  Phantom Calls? n


                                ASAI PROPRIETARY FEATURES

                                       Agent States? n
```

**Figure 25: Maximum IP Registrations by Product ID screen**

```
                                                                Page 9 of x
                    MAXIMUM IP REGISTRATIONS BY PRODUCT ID

Product ID_Rel. Limit     Product ID_Rel. Limit     Product ID_Rel. Limit
__________ ___. _____     __________ ___. _____     __________ ___. _____
__________ ___. _____     __________ ___. _____     __________ ___. _____
__________ ___. _____     __________ ___. _____     __________ ___. _____
__________ ___. _____     __________ ___. _____     __________ ___. _____
__________ ___. _____     __________ ___. _____     __________ ___. _____
__________ ___. _____     __________ ___. _____     __________ ___. _____
__________ ___. _____     __________ ___. _____     __________ ___. _____
```

**Figure 26: IP Network Region screen**

```
change ip-network-region n                                Page 1 of x
                            IP NETWORK REGION
  Region: n
Location:                 Authoritative Domain:
    Name:
                               Intra-region IP-IP Direct Audio: n
MEDIA PARAMETERS               Inter-region IP-IP Direct Audio: n
   Codec Set: 1                           IP Audio Hairpinning? n
UDP Port Min: 2048
UDP Port Max: 3028                    RTCP Reporting Enabled? y
                                RTCP MONITOR SERVER PARAMETERS
DIFFSERV/TOS PARAMETERS         Use Default Server Parameters? y
 Call Control PHB Value:                    Server IP Address:   .   .   .
        Audio PHB Value:                          Server Port: 5005
        Video PHB Value:
802.1P/Q PARAMETERS                  RTCP Report Period(secs): 5
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 7
                                     AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                    RSVP Enabled? y
  H.323 Link Bounce Recovery? y               RSVP Refresh Rate(secs): 15
 Idle Traffic Interval (sec): 20      Retry upon RSVP Failure Enabled? y
   Keep-Alive Interval (sec): 6                       RSVP Profile:
            Keep-Alive Count: 5       RSVP unreserved (BBE) PHB Value: 40
```

### Figure 27: IP Network Region screen

```
change ip-network-region n                                  Page 2 of x

                                IP NETWORK REGION

INTER-GATEWAY ALTERNATE ROUTING/DIAL PLAN TRANSPARENCY
Incoming LDN Extension:
Conversion to Full Public Number - Delete: ___Insert: ___
 Maximum Number of Trunks to Use for IGAR:
Dial Plan Transparency in Survivable Mode? n


BACKUP SERVERS IN PRIORITY ORDER          H.323 SECURITY PROCEDURES
1                                         1
2                                         2
3                                         3
4                                         4
5
6                                        Allow SIP URI Conversion? y

TCP SIGNALING LINK ESTABLISHMENT FOR AVAYA H.323 ENDPOINTS
   Near End Establishes TCP Signaling Socket? y
                      Near End TCP Port Min: 61440
                      Near End TCP Port Max: 61444
```

### Figure 28: Inter Network Region Connection Management screen

```
change ip-network-region n                                  Page   3 of x
                  Inter Network Region Connection Management
 src dst codec direct   Total          Video                          Dyn
 rgn rgn  set  WAN   WAN-BW limits Norm Prio Shr Intervening-regions   CAC IGAR
 3   1    1       y    256:Kbits             y                          n
 3   2    1       n       :NoLimit           n                          n
 3   3    1               :NoLimit                                      n
 3   4    1       n       :NoLimit           n                          n
 3   5    1       n       :NoLimit           n                          n
 3   6    1       y       :NoLimit           y                          n
 3   7    1       y     10:Calls             y                          n
 3   8
 3   9
 3   10
 3   11
 3   12
 3   13
 3   14
 3   15
```

**Note:**

On the COR (Class of Restriction) form, be sure the Paste field defaults to “n” (no). Undesirable actions can occur if this field is defined. This requirement applies to both the station extension and the loginID extension.

**Figure 29: Class of Restriction screen (page 1)**

```
change cor n                                                  Page 1 of x
                            CLASS OF RESTRICTION
                COR Number: n
           COR Description: supervisor

                       FRL: 0                                  APLT? y
  Can Be Service Observed? n                Calling Party Restriction: none
Can Be A Service Observer? y                 Called Party Restriction: none
 Partitioned Group Number: 1              Forced Entry of Account Codes? n
         Priority Queuing? n                        Direct Agent Calling? y
     Restriction Override: none              Facility Access Trunk Test? n
     Restricted Call List? n                       Can Change Coverage? n
   Unrestricted Call List? __ __ __ __ __ __ __ __ __ __
            Access to MCT? y                  Fully Restricted Service? n
Group II Category For MFC: 7                  Hear VDN of Origin Annc.? n
         Send ANI for MFE? n_                  Add/Remove Agent Skills? y
            MF ANI Prefix: ______              Automatic Charge Display? n
Hear System Music on Hold? y     PASTE(Display PBX Data on telephone)? n
                            Can Be Picked Up By Directed Call Pickup? n
                                       Can Use Directed Call Pickup? n
                                       Group Controlled Restriction: inactive
```

**Figure 30: Class of Restriction screen (page 2)**

```
change cor nn                                                 Page 2 of x
                        CLASS OF RESTRICTION

                     MF Incoming Call Trace? n
                Brazil Collect Call Blocking? n
                     Block Transfer Display? n
Block Enhanced Conference/Transfer Displays? y
                     Remote Logout of Agent? n

                           Station Lock COR: 10
  Outgoing Trunk Disconnect Timer (minutes):
       Line Load Control:
 Maximum Precedence Level:        Preemptable?
 MLPP Service Domain:
      Station-Button Display of UUI IE Data?
      Service Observing by Recording Device?
                         ERASE 24xx USER DATA UPON
        Dissociate or unmerge at this phone: none
          EMU login or logoff at this phone: none
           Mask CPN/NAME for Internal Calls:
```

**Figure 31: Class of Restriction screen (page 3)**

```
change cor nn                                              Page 3 of x
                     CLASS OF RESTRICTION

            SAC/CF Override by Team Btn? n
SAC/CF Override Protection for Team Btn? n
```

**Figure 32: Class of Restriction screen (page 4)**

(Use pages 4 to 13 to assign up to 995 CORs)

```
change cor nn                                              Page 4 of x
                     CLASS OF RESTRICTION
CALLING PERMISSION (Enter y to grant permission to call specified COR)
  0? n     15? n     30? n     44? n     58? n     72? n     86? n
  1? n     16? n     31? n     45? n     59? n     73? n     87? n
  2? n     17? n     32? n     46? n     60? n     74? n     88? n
  3? n     18? n     33? n     47? n     61? n     75? n     89? n
  4? n     19? n     34? n     48? n     62? n     76? n     90? n
  5? n     20? n     35? n     49? n     63? n     77? n     91? n
  6? n     21? n     36? n     50? n     64? n     78? n     92? n
  7? n     22? n     37? n     51? n     65? n     79? n     93? n
  8? n     23? n     38? n     52? n     66? n     80? n     94? n
  9? n     24? n     39? n     53? n     67? n     81? n     95? n
 10? n     25? n     40? n     54? n     68? n     82? n     96? n
 11? n     26? n     41? n     55? n     69? n     83? n     97? n
 12? n     27? n     42? n     56? n     70? n     84? n     98? n
 13? n     28? n     43? n     57? n     71? n     85? n     99? n
 14? n     29?
```

**Figure 33: Signaling Group screen**

```
display signaling-group 7
                               SIGNALING GROUP
Group Number: 7              Group Type: sip
                       Transport Method: tls

Near-end Node Name: clan-1a02             Far-end Node Name: ses
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                      Far-end Network Region: 1
       Far-end Domain: trlvri.transcomeurope.com

                                           Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload          Direct IP-IP Audio Connections? y
                                                  IP Audio Hairpinning? n
         Enable Layer 3 Test? n
 Session Establishment Timer(min): 3
```

**Figure 34: Trunk Group screen (page 1)**

```
display trunk-group 7                                    Page   1 of  21
                              TRUNK GROUP

Group Number: 7                    Group Type: sip          CDR Reports: y
  Group Name: Sip Enablement Server       COR: 1       TN: 1        TAC: 287
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                      Signaling Group: 7
                                                    Number of Members: 60
```

**Figure 35: Trunk Group screen (page 2)**

```
display trunk-group 7                                    Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name? y

                                          Redirect On OPTIM Failure: 5000

            SCCAN? n                               Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 600
```

**Figure 36: Trunk Group screen (page 3)**

```
display trunk-group 7                                    Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                    Maintenance Tests? y


                     Numbering Format: private
                                        UUI Treatment: service-provider

                                          Replace Restricted Numbers? n
                                         Replace Unavailable Numbers? n


               Send UCID? y


 Show ANSWERED BY on Display? y
```

**Figure 37: Trunk Group screen (page 4)**

```
display trunk-group 7                                    Page   4 of  21
                            PROTOCOL VARIATIONS

                      Mark Users as Phone? y
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n

             Telephone Event Payload Type:
```

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